

Sound Lab[®]
TDS
Software User's Manual

Crown International, Inc.
1718 W. Mishawaka Road
Elkhart, IN 46517-4095

©1997 Crown International, Inc.
102842-1
6/97

Trademarks

The Techron TEF System 20 is manufactured by TECHRON Division of Crown International, Inc., Elkhart, Indiana, U.S.A., under license from Jet Propulsion Laboratories, California Institute of Technology, Pasadena, California, U.S.A.

TEF*, TEF System 20*, and TECHRON* are registered trademarks of TECHRON Division of Crown International, Inc., Elkhart, Indiana.

The trademark Sound Lab* is licensed under United States trademark registration number 1,424,678.

Apple* and Macintosh* are registered trademarks of Apple Computer, Incorporated.

IBM and all IBM products mentioned in the manual are registered trademarks of International Business Machines, Incorporated.

Motorola* is a registered trademark of Motorola, Incorporated.

MS-DOS* and Microsoft* are registered trademarks of Microsoft Corporation.

The Techron TEF System 20 incorporates technologies requiring validated licensing upon export from the United States in accordance with U.S. Export Administration Regulations.

Acknowledgment

Techron gratefully acknowledges the kindness of author and publisher in giving permission to reproduce their materials in *Sound Lab for PC Software User's Manual*. Howard W. Sams & Co.: "Basics of Sound," from *Introduction to Professional Recording*, by Bruce Bartlett, ©1987.

TECHRON is a division of Crown International, Inc., 1718 West Mishawaka Road, Elkhart, Indiana, 46517-4095.

***Dedication: "It is the theory which
decides what we can
observe."***

In 1967, a remarkable man, Richard Heyser, brought a new paradigm to the science of sound. His passion for the study of acoustics, sound perception and audio measurement launched him upon a solitary journey in uncharted areas between paradoxes in the understanding of sound.

His travels led from one end of the spectrum of language and thought to the extreme opposite. Here was the problem: Things don't always sound the way they measure. While statistical excellence could be described in the symbolic language of objective analysis, it was often contradicted by the language descriptions of subjective perception. And it was in this world of paradox that Dick Heyser navigated comfortably until he found an entirely new world emerging on the horizon.

The expansive Heyser often quoted Einstein: "It is the theory which decides what we can observe." From the center of that theory, Heyser began to push into new horizons of sound measurement, unrestricted by the boundaries of convention.

Along the way, his observations, drawn from nature itself, fueled his thinking into even deeper areas of understanding with far-reaching implications. He generously shared the results of his explorations at a

Syn-Aud-Con gathering in 1984 with the following preface.

1. Nature proceeds without prejudice to the way you look at it.
2. There are an infinite number of equally valid ways of looking at nature—which he named the principal of alternatives.

The background of his work brought him to question: If there is no preferred way of observing any one event, was there a way, perhaps, to travel from one alternative frame of reference to another?

The experimental Heyser, comfortable with the validity of both the subjective and objective domains of sound evaluation, began seeking maps that would enable traveling between both frames of reference.

He reasoned: If there was a way, and the event still retains the same essence, what essentially doesn't change as you transfer from one alternative to another? Heyser's pursuit of the dynamics and mathematics underlying this assumption led eventually to its application in sound analysis which Heyser named "time delay spectrometry"—a specific class of integral transform that maps among domains of different dimensionality. Indeed it was "a small tear on the fabric of contemporary understanding" of electroacoustic measuring.

Heyser lived to see the beginnings of the acceptance and incorporation of his thinking into sound analysis equipment, and its actual application in the fields of audio, medicine and geophysics. Unfortunately, long before he could transfer the abundance of all his thought into realities, he died suddenly in 1987, carrying much unfinished business with him.

A man before his time, Heyser, some have said, was the Newton of our age. It remains to be seen, as time delay spectrometry continues to grow with better equipment and software to make practical use of Heyser's theory. The Techron division of Crown International takes particular honor in its good fortune of being one avenue in which the thinking of Richard Heyser is becoming available in useful terms to people solving problems in sound analysis.

We therefore dedicate this manual to the memory of Richard Heyser, the remarkable man who gave us a new theory in which to not only observe, but to continue to work, as well.

Richard C. Heyser received his B.S.E.E. degree from the University of Arizona in 1953. Awarded the AIEE Charles LeGeyt Fortescue Fellowship for advanced studies, he received his M.S.E.E. from the California Institute of Technology in 1954. The following two years were spent in post-graduate work at Cal Tech leading toward a doctorate. During the summer months of 1954 and 1955, Mr. Heyser was a research engineer specializing in transistor circuits with the Motorola Research Laboratory, Phoenix, Arizona. From 1956, until his death in 1987, he had been associated with the California Institute of Technology Jet Propulsion Laboratory in Pasadena California where he was a senior member of the JPL Technical Staff.

Contents

Trademarks Acknowledgment Dedication

Section 1: Introduction	1-1
Unpacking	1-1
Typical equipment for acoustical measurements.....	1-2
What you need to know	1-2
About this manual	1-2
Customer support	1-5
 Section 2: Sound Lab TDS Install program	 2-1
Copyright screen	2-2
Select a COM port.....	2-3
Check install selections	2-5
File transfer	2-5
Required files	2-6
 Section 3: Navigation in Sound Lab	 3-1
Navigation in Sound Lab	3-1
Getting around	3-2
Selecting a menu title from the main menu	3-3
Exiting the menu	3-3
Selecting menu items	3-4
Entering information in a sub-menu	3-5
How to undo what you entered	3-6
Helpful features for working with the software	3-7
Moving from module to module	3-8

Section 4: Menus for the TDS module

File menu	4-1
Measure menu	4-11
Parameters menu	4-15
Setting test parameters	4-15
Time Response (ETC) Parameters	4-16
Frequency Response	4-20
Noise Test (NC)	4-27
STI/RASTI (STI)	4-29
3D Test (3D)	4-30
Averaging	4-36
Output	4-39
Display pull-down menu	4-41
Time Response...Magnitude	4-44
Time Response...Heyser Spiral	4-46
Frequency Response	4-47
Frequency Response...Phase	4-48
Frequency Response...Magnitude	4-49
Frequency Response...Magnitude and Phase	4-50
Frequency Response...Nyquist	4-51
Frequency Response...Heyser Spiral	4-53
3D (Waterfall)	4-55
NC (Noise)	4-57
STI	4-58
Other Display options	4-59
Summary of Display menu options	4-69
Input menu	4-73
Settings sub-menu	4-74
Calibration sub-menu	4-76
Calibrating the display	4-76
Communication sub-menu	4-81

Section 5: Performing measurements	5-1
Introduction	5-1
To make Sound Lab TDS tests	5-1
Performing a time response test on a loudspeaker	5-2
Performing a frequency response test	5-6
Noise criteria measurements	5-9
Equipment needed	5-9
NC measurements with a microphone	5-9
NC measurements with sound level meters	5-10
Introduction to STI	5-13
STI and RASTI measurements	5-13
Performing the measurement with calibrated talker/speaker	5-15
Equipment arrangement for calibration	5-15
Setting up parameters	5-16
Arrange the equipment for STI	5-19
Driving the system directly	5-20
Adding noise to STI measurements	5-21
Evaluating the STI graph and summary table	5-22
STI summary table display	5-23
About the STI measurement process	5-25
Introduction to 3D measurements	5-27
Measuring loudspeakers with 3D	5-28
Measuring rooms with 3D	5-30
Overlap ratio in 3D displays	5-31
Section 6: Practice measurements	6-1
Two basic tests	6-1
About the ETC	6-1
Reading an ETC display	6-3
Doing a Time Response Test (ETC)	6-4
Equipment arrangement	6-4
Setting the input parameters	6-5
Calibrating the display	6-6
Setting the parameters for the Time Response test	6-7
Setting up the screen display	6-8

Running the Time Response test	6-9
The power of the data cursor	6-10
Using the cursor to examine the data	6-11
Finding reflections	6-12
Doing a Frequency Response Test (TDS)	6-14
Setting the parameters for the Frequency Response Test	6-15
Setting the screen display	6-17
Running the test.....	6-18
Adjusting the receive delay to tune in to the phase	6-19
Making %ALcons and RT60 measurements	6-25
Terms to know	6-25
An appropriate ETC.....	6-26
ETC display example	6-27
Parameters	6-28
Power of the cursor	6-28
Additional information	6-30
Classical RT60	6-31
Early Decay Time	6-33
Finding the %ALcons	6-34

Appendix A: Basics of sound

A-1

Waves	A-1
Illustration A-3 Three cycles of a wave.	A-3
Characteristics of sound waves	A-3
Amplitude	A-3
Wavelength	A-4
Phase and phase shift	A-4
The phase of various points on a wave.	A-5
Harmonic content	A-6
Behavior of sound in rooms	A-7
Echoes	A-7
Reverberation	A-9
Reverberation time	A-11
Room modes	A-11



Appendix B: How TEF measurements work **B-1**

Reflections can be kept out of measurement	B-2
Relationship between Time/Frequency	B-3
Fourier transform	B-4
Measurement resolution	B-5
Trade-off between time and frequency	B-7
Interrelated parameters	B-8
Sweep rate	B-8
Frequency span	B-9
Receive (signal) delay	B-9
Bandwidth/Sweep rate	B-10
Space window considerations	B-12

Appendix C: Data interpretation **C-1**

Is it reasonable?	C-1
Is it repeatable?	C-2
Some things to watch for	C-4

Appendix D: On-screen messages **D-1**

Appendix E: ASCII file formats **E-1**

Header block format	E-1
Data block format	E-1
ETC file format	E-2
The ETC header	E-2
The ETC data	E-2
Notes on ETC files	E-3
NC ASCII files	E-6
The NC header	E-6
The NC data	E-6
Notes on NC files	E-7

The TDS header	E-9
The TDS data	E-9
Notes on TDS files	E-11
STI and RASTI file formats	E-13
STI analyzer settings block	E-13
STI summary table block	E-13
STI ETC header block	E-14
STI ETC data blocks	E-14
Notes on STI summary table block	E-18
Notes on STI noise data block	E-20
Notes on STI ETC block	E-22
3D file format	E-25
The 3D master file header	E-25
Notes on 3D files	E-26

Appendix F: Glossary	F-1
-----------------------------	------------

Appendix G: Best Frequency Resolution and the TEF Resolution V	G-1
---	------------

Bibliography

Section 1: Introduction

Sound Lab TDS uses the measurement technique of Time Delay Spectrometry (TDS) to make Time, Energy and Frequency (TEF) measurements. Time Delay Spectrometry is a time selective measuring technique suited to making transfer-function measurements on devices that have a well-defined input and output.

TDS falls into the general class of two-port measurement methods where the test system generates a test signal, sends the test signal to the device under test, and gathers, analyzes and displays data about the output relative to the input of the device. *Sound Lab TDS* measures traditional parameters such as frequency response and phase response, along with a number of TDS specific measurements such as time and distance and energy-time-frequency curves.

Sound Lab TDS software displays and stores data on a variety of microcomputers but requires the Techron TEF 20 analyzer to make measurements and collect data.

Unpacking

Sound Lab TDS software contains the following items:

- User's manual
- 1.44M, 3.5" Distribution disk
- Function key template

Typical equipment for acoustical measurements

In addition to *Sound Lab TDS* software you may need the following equipment:

- MS-DOS computer (20MHz 386 with math coprocessor)
- TEF 20 analyzer
- Microphone
- Power amplifier
- Speaker
- Hard drive or formatted disks for storing data

What you need to know

This manual assumes your familiarity with (1) MS-DOS and utilities that came with your computer, (2) your mouse and its installation and operation, and (3) general acoustics and sound system design.

About this manual

This manual is a reference manual for *Sound Lab TDS* and is not intended to be a tutorial on how to make acoustic measurements. The remainder of the manual contains sections to help you start using *Sound Lab TDS*.

Section 1: Introduction to *Sound Lab TDS*

What you purchased, what you need know, equipment you need and where to get help.

Section 2: Installing *Sound Lab TDS*

Step by step instructions to install *Sound Lab TDS* on your computer.



Section 3: Navigation in Sound Lab TDS

How to move around in *Sound Lab TDS*. Explains menus, important keys and moving around within the program and from module to module.

Section 4: Menus for the TDS module

These menus are detailed: *File*, *Measure*, *Parameters*, *Display*, and *Input*.

Section 5: Performing measurements

This section of the manual details the sequence used to make time and frequency response tests, noise tests, STI/RASTI tests, and 3D tests.

Section 6: Practice measurements

A practice measurement session you can set up to perform the most fundamental TEF measurements — time response and frequency response. Explanation and exercise on cursor functions for % ALcons and RT60 calculations.

Appendix A: Basics of sound

Fundamental acoustic principles and basic concepts you need to know to understand TEF measurements.

Appendix B: How the TEF analyzer works

This section describes how the TEF analyzer works, discusses the relationship between time and frequency, measurement resolution, the interrelated parameters and space window considerations.

Appendix C: Data interpretation

Described and explained are several tests to help determine the legitimacy of a measurement along with common errors that can creep into measurements.



Appendix D: On-screen messages

Appendix E: File structures

Appendix F: Glossary of terms

Appendix G: TEF resolution V



Customer support

All Techron products are backed by a customer support system. If you need assistance beyond that provided in the manual, follow these steps:

- Try to duplicate the problem, keystroke by keystroke, to see exactly what was done.
- Have the date and version number of software available. This information is displayed in the *File...About* sub-menu.
- Have the date and version number of the manual available (found on the front cover page).
- Be at the TEF analyzer and call customer support.

Customer support for Techron hardware and software can be reached by dialing:

(800) 833-8575 North America
(219) 294-8295 Outside North America
(219) 294-8302 FAX

Techron Customer Support takes calls daily from 8:00 AM to 5:00 PM Eastern Standard Time. Before 8:00 or after 5:00, you may leave a recorded message via voice mail or send a message via FAX. The automated TEF FAX Information System has an index of various topics related to the TEF analyzers and software that you can request to have transmitted to your FAX machine.



Section 2: Sound Lab Install program

Assumed is a minimal knowledge of DOS and that you know how to:

- turn on your machine
- come to a prompt
- format a disk
- make a backup copy
- change directories

If you are not sure how to perform these tasks, refer to your DOS manual before starting the *Sound Lab Install* program.

Your TEF *Sound Lab* package includes all the files and programs necessary for installation. While *Sound Lab* works on both hard drive and floppy-based systems, a hard drive is recommended.

Before you install this *Sound Lab* software, make backup copies of your disks and store originals in a safe place. Install from the copy which is not write protected. (This means it can be written to.)

Let's assume that you're installing *Sound Lab TDS* on your hard disk, which is drive C, from floppy drive A.

Put the *copy disk* in drive A and type *A:TDSINST*. Press *Enter* to continue.

SOUND LAB - TDS **Installation Program 2.0**

Copyright (c) 1991,94
TECHRON Division of Crown International, Inc.
P.O. Box 1000
1718 W. Mishawaka Rd.
Elkhart, IN 46515-1000
(219) 294 - 8295

ENTER - Continue

ESC - Return to Dos

Copyright screen

The copyright screen is first screen you'll see in the *Sound Lab Install* software. Press *Enter* to continue or *Esc* to return to DOS.

The second screen asks you to name the *destination drive:\path* where you will install the program. We recommend that you choose the default drive and directory, *C:\SL*.

Enter destination drive:\path

C:\SL\

ENTER - Continue

ESC - Return to DOS

Assuming you wish to proceed, press *Enter*.

A third screen appears if *TDS.SET* is not found.



To select a COM port

type the number of the desired COM port
or H for the Host Interface option
[1, 2 or H]

HI

Check to be sure TEF-20 is connected to this port

ENTER - Continue

ESC - Back Up

Select a COM port

This screen tells you to type the number of the desired *COM port (1 or 2)* or *H* to select the *Host Interface* and a message prompting you to:

"Check to be sure the TEF 20 is connected to this port."

(The TEF does not necessarily need to be connected when you're simply installing the software.)

Note: If you intend on using parallel communications select *Host Interface* at this time. You will be able to select parallel communications once the program is installed.

Press *Enter* to continue.



If you selected 1 or 2, you will proceed immediately to a screen with a message:

Copy the demo files? (Y/N)

Y/N - Continue

ESC - Back Up

"Copy the demo files?"

Choose *Y* or *N* to continue. We recommend that you choose "yes."

If you selected the *H* option, a new screen will appear with the following message:

Use the left & right arrow keys

to highlight the HI base address as set by
the DIP switches on the HI interface board.

108 148 188 1C8 208 248 288 2C8 **308** 348 388

ENTER - Continue

ESC - Back Up

"Use the left and right arrow keys to highlight the HI base address as set by the DIP switches on the HI interface board."

108	148	188
1C8	208	248
288	2C8	308
348	388	

The default base address is 308. Make your choice, then press *Enter*.

The following message appears:

"Copy demo files?"

Choose *Y* or *N* to continue. We recommend that you choose "yes."

Another screen will appear, asking you to confirm your selections.

Check install selections

Source = A:\SL

Destination = C:\SL

HI

Base Address = 308

Copy demo files

ENTER - Continue

ESC - Back Up

Check install selections

This screen allows you to review the installation choices you've made so far. If you wish to make any changes at this point, press *Escape* to back up to the screen where you need to make the change.

If the information appears to be correct, press *Enter*.

File transfer

The final screen announces that the *File Transfer* is taking place. The *Install* program starts transferring the files within the directories that were created, on the drive you've selected.

This install program will only transfer files, and will not change existing *CONFIG.SYS* or *AUTOEXEC.BAT* files.

As soon as all files are transferred, you will be given the message:

"TDS has been installed. Type SL to begin using Sound Lab TDS software."

Sound Lab will open in the last module of *Sound Lab* you used. Choose the *Go To* option under the *File* menu to select *TDS* to open *Sound Lab TDS*.

Required files

After installation, use the MS-DOS command *Tree* to confirm that the following files are in the *SL* directory. All of the files shown in the following illustration must be present, but not necessarily in the order shown.

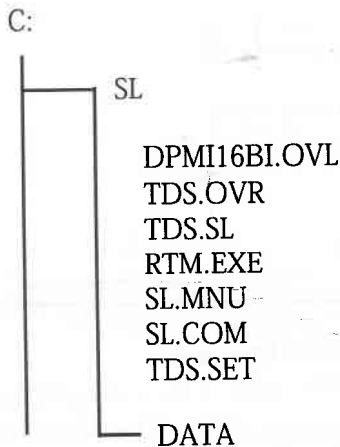
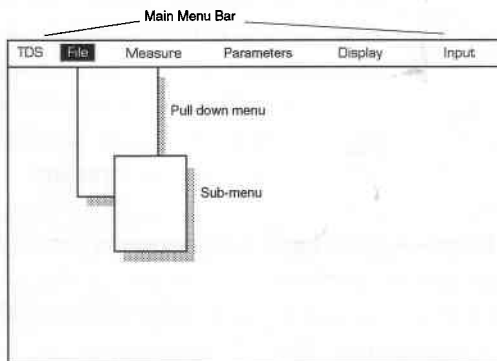


Illustration 2-1
The directory tree after TDS installation

Section 3: Navigation in Sound Lab

Navigation in Sound Lab

Sound Lab TDS software runs under DOS and uses a mouse input device and pull-down menus and sub-menus, which contain either parameters or commands. In this manual the terms “*command*”, “*parameter*,” “*option*” and “*item*” are used interchangeably to refer to menu selections. The current *Sound Lab TDS* module name appears in the upper left hand corner, along with the major menus in the *Sound Lab TDS* program: *File*, *Measure*, *Parameters*, *Display*, and *Input*.



Pull-down menu—To run *Sound Lab TDS*, you select one of the five menu titles from the main menu bar. *Sound Lab TDS* then drops down the rest of the menu, displaying the list of items on the menu. One or more sub-menus allow you specify your instructions about test parameters.

Illustration 3-1
Main Menu bar



Getting around

Sound Lab TDS offers you three ways to navigate through the menus and sub-menus: a single-button *Mouse*, the *Arrow keys*, and *Quick keys*.

Mouse

You must have installed the device driver that controls the mouse to be able to use it in *Sound Lab TDS*. If you do not have a mouse driver installed, see your mouse manual for installation procedures. *Sound Lab* uses the left button of the mouse to select items and move the cursors in displays.

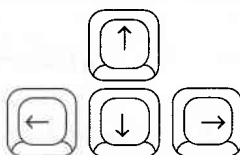


Illustration 3-2
Arrow Keys

Arrow keys—The Arrow keys are a group of four keys with imprinted direction arrows. Press the *Left* and *Right Arrow* keys to navigate along the main menu bar displaying each menu title. The *Up* and *Down Arrow* keys allow you to navigate up and down between sub-menu options.

Quick keys—Quick keys are keyboard letter keys that are shortcuts to performing actions in the software. Quick keys are displayed in a color different than the rest of a command. Not all commands are available through their quick-keys under every condition.



Selecting a menu title from the main menu

There are three ways to select a menu title from the main menu bar.

Mouse—Point and click the menu title with the mouse.

Arrow keys—Press the *Right Arrow* key or *Left Arrow* key to highlight your choice, then press *Enter* to select it.

Quick keys—Type the highlighted letter in the menu title (such as the *D* in *Display*.)

Exiting the menu

No matter how you select a menu, you can exit the menu (close the window) by selecting *Alt-Close* in any of the three ways:

Mouse—Point and click *Alt-Close*.

Arrow keys—Press the *Down arrow* key to *Alt-Close* and press *Enter*.

Quick keys—Hold *Alt* and press *C*.

Selecting menu items

After you select a menu title, a sub-menu appears with a list of commands on the left and their settings on the right.

Selecting an item will invoke one of three things:

- A new sub-menu
- A highlighted area called an *edit field*.
- A toggle or switch between two settings.

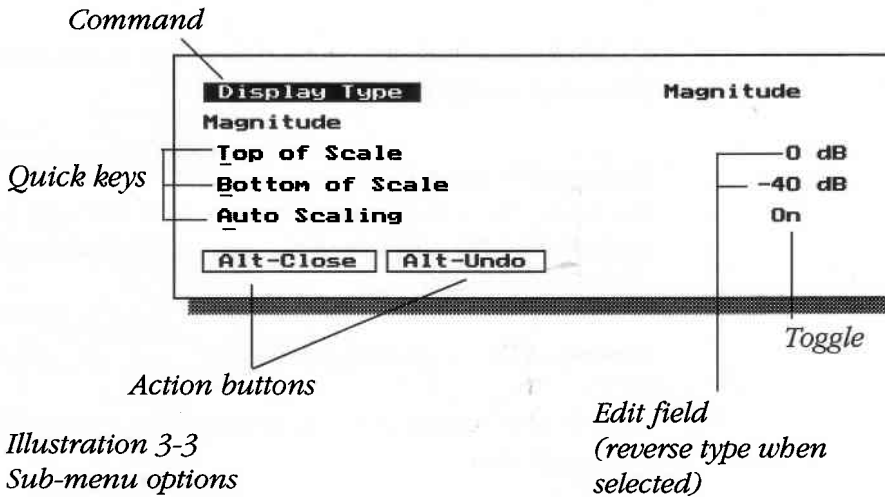


Illustration 3-3
Sub-menu options

Entering information in a sub-menu

Choose the item you want with the mouse, Arrow keys, or highlighted letter.

Reference Unit	pascals
Volts Per Reference Unit	0.00226
Zero dB Reference Value	0.00002
Propagation Speed	1130.0
Distance Unit	feet
<input type="button" value="Alt-Close"/> <input type="button" value="Alt-Undo"/>	

Edit field—some items require you to type information in an edit field. When you select one of these commands, a space (edit field) is highlighted where you can enter numbers or text from the keyboard.

Illustration 3-4
Edit field.

It is not necessary to erase any highlighted numbers or text. When you begin typing, the new data replaces the old. To edit information in the highlighted field, you may press the *Left Arrow* key to navigate to the left in the string. The *Home* key will jump to the beginning of the string. Press the *Right Arrow* key to navigate to the right in the string. The *End* key will jump to the right end of the string. Then, press *Enter* to store the settings.

When you're setting a series of parameters, it's more convenient enter a setting, then press the *Down Arrow*. This enters the setting and tabs down to the next edit field. Then you can type in the next setting, and so on.

If you enter a setting that the program can't use, the closest valid setting will be displayed. To change a setting you typed, re-select the edit field and re-enter the data. To cancel an edit you just made, press *ESC* while the edit field is highlighted.



Toggle choices—If a command provides a choice between two options, a toggle switches between the two alternate choices. To switch between the two choices, re-select the command. Or you can toggle the setting by clicking on the setting with the mouse.

How to undo what you entered

Many sub-menus have a command called *Alt-Undo*. Choosing *Alt-Undo* resets all the sub-menu parameters to the settings they had when the current sub-menu was opened. The *Undo* feature works only until the sub-menu window is closed. You cannot reopen the window and undo those changes.

To use *Alt-Undo*:

- Using your mouse, point and click on *Alt-Undo*.
- Press the *Down Arrow* key to highlight *Alt-Undo*, then press *Enter*.
- Hold down the *Alt* key and press *U*.



Illustration 3-5
Alt-Close action
button

Helpful features for working with the software

Action buttons—Usually appear at the bottom of a menu or sub-menu and cause an immediate action to occur when chosen. Some of the action buttons common throughout the menus are:

Alt-Close—Accepts the parameters as presented and closes the window.

Alt-Undo—Returns all settings to the original settings in the window.

Function keys—The *Function keys* (F1-F10) allow you to enter many commands without selecting a menu. Function keys may be used in conjunction with the *Control*, *Shift* or *Alternate* keys.

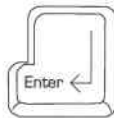


Illustration 3-6
Entry cursor

Enter key—Press the *Enter* key at the end of numeric entries such as entering sweep rate or start and stop frequencies. When you press *Enter*, the program accepts your entry, and you can continue with another command on the menu.



Illustration 3-7
Escape key

Escape key—In numeric or typed entries, press the *Escape* key, *Esc*, to cancel a data entry and return to your previously entered data. The *Escape* key is also used to cancel a test in progress. Note that the *Escape* key does not return you to the previous menu. The *Alt-Close* key combination returns you to the previous menu or to the menu bar.

Message boxes—Pertinent information to report the status of a process or calculation is presented in windows called *message boxes*. Message boxes contain information only. They typically appear at the top center of the screen under the main menu bar.

Moving from module to module

Sound Lab software is comprised of several modules that make up a total measurement package. When you start *Sound Lab TDS* for the first time, the TDS module is loaded. You choose other modules with the *Go to* command in File menu. As you install new modules, they will appear in the *Go to* sub-menu.

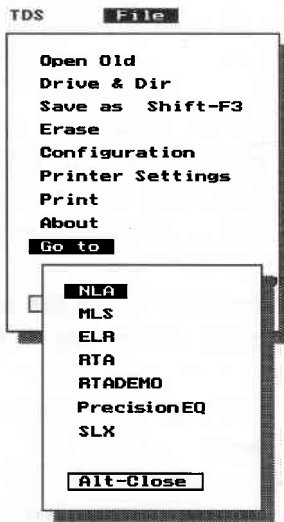


Illustration 3-8
Go to sub-menu

Section 4: Menus for the TDS module

TDS

File**Open Old****Drive & Dir****Save as Shift-F3****Erase****Configuration****Printer Settings****Print****About****Go to****Quit****Alt-Close**

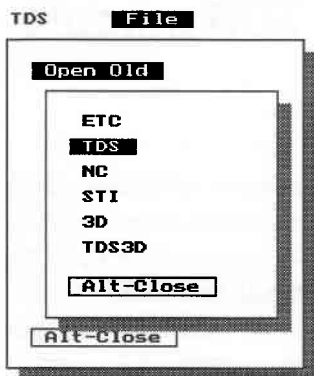
File menu

Sound Lab communicates with disk drives and printers, and accesses other modules under the *File* menu. Four commands (*Open Old*, *Drive & Dir*, *Save As*, and *Erase*) perform tasks for data storage, retrieval and path navigation. *Configuration* provides a way to load, save and erase configuration files. *Printer Settings* and *Print* print the data on the printer.

About displays the current version of the software, and if you are connected to the TEF analyzer, will show the EE and EEPROM version numbers and date of software. The

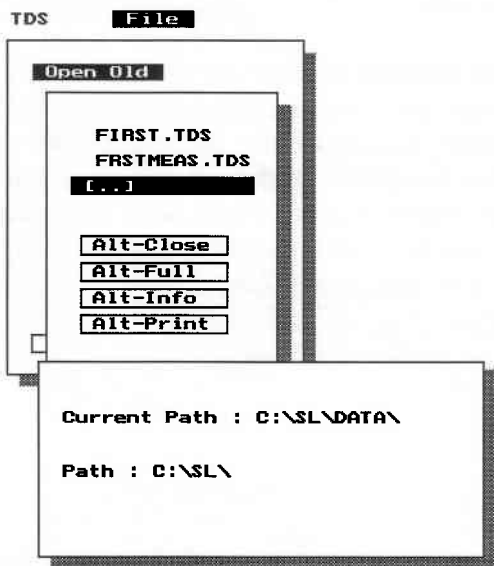
Go to command allows you to access other modules that have been installed. *Quit*, the last command, is the only proper way to exit *Sound Lab*.

Illustration 4-1
File menu



Open Old...The *Open Old* command retrieves and displays stored data and allows you to navigate to other directories.

When you choose *File, Open Old*, a list box appears showing available file types from which to choose. After you choose a file type, a list box opens displaying filenames and/or directories and subdirectories. Filenames appear as a name plus extension, e.g. *FRSTMEAS.TDS*. A directory or subdirectory has no extension and is enclosed by brackets, e.g., *[DATA]*.



The *[.]* represents the parent directory. When it is highlighted, press *Enter* or double click the mouse to change the path to the parent of the current directory or other directories.

No more than ten items can appear in the list box at any time. The word *(more)* will appear at the bottom of any list greater than ten. When you choose this option, the list advances to show more items.

Note: The commands *Erase* and *Configuration* use the same or similar routine for file path navigation and file selection.

Illustration 4-2 File...Open Old sub-menus.



Four other commands are displayed:

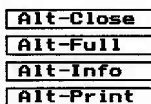


Illustration 4-3
Action buttons

Alt-Close—Closes the sub-menu without making a selection.

Alt-Full—Expands the file information box showing files and directories with details such as file size, date and time created. *Alt-Brief* appears in the file information box when you choose *Alt-Full*. Choosing *Alt-Brief* collapses the file information box to a list box of file or directory names only. If you want to revert to the file information box, select *Alt-Full*.

Alt-Info—If a filename is highlighted, *Alt-Info* opens a file information box containing the job description about that file. If a directory name is highlighted, the information box will show the current path and the path you can travel to if you press *Enter*.

Alt-Print—Selecting *Alt-Print* will print the job description that was recorded with the *Save As* command for each of the files listed in the sub-menu. Seven job description entries will print per page.

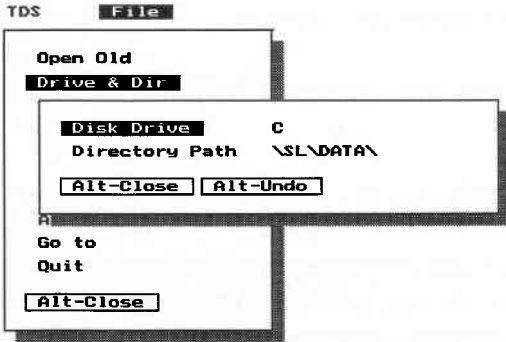


Illustration 4-4
Drive & Directory sub-menu

Drive & Dir - DOS allows you to group files in directories. The *Drive & Dir* command allows you to select a drive and directory for data storage or retrieval. *Sound Lab TDS* defaults to drive *C:* and directory *\\SL\\DATA*.

Note: The Drive and Directory will reflect any changes made when the *File...Open Old* command is used.

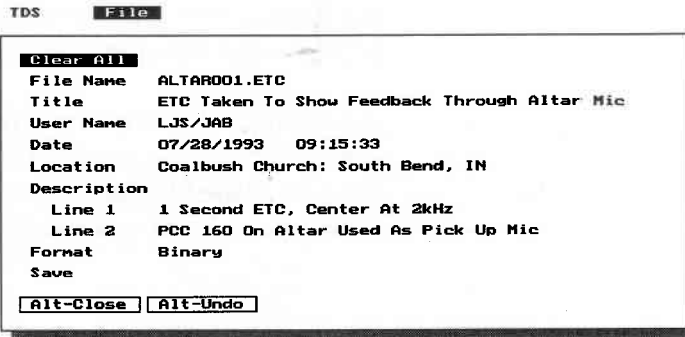


Illustration 4-5 Save as sub-menu

Save as—Saves the current test data, settings, parameters and job description to the drive and directory selected in the *Drive & Dir* sub-menu. Until you execute *Save as*, test data is stored only in memory and will be erased if a new measurement is performed.

Information that appears in the *Save as* sub-menu is carried over from the last measurement, re-called file or configuration file.

The choices in the *Save as* sub-menu are:

Clear All: Erases all the entries made in the *Save As* sub-menu, and changes the file name to *noname*.

Additional data such as test *Title*, *User Name*, and *Date* and *Location* of the measurement are optional. The *Date* is entered automatically by *Sound Lab*, but you may change it. You may also enter two lines to describe the measurement under *Description*. *Format* allows you to save files in binary or ASCII file format. Each file format has advantages and disadvantages.

- A binary file takes up less space on the disk and is faster to open and display. It is not portable to spreadsheets.
- ASCII data requires several times more disk space and takes longer to open and display, but you may use other programs, such as spreadsheets, to view and manipulate the data.

Note: The format of the file may be changed at any time, providing it was saved. Use *Open Old* to load the data into memory, then change the file format and resave either under the same or a new name.

Save +1—This command is only available by pressing *F3*. *Save +1* will cause *Sound Lab* to add *001* to the end of the current measurement file name. The displayed data will be saved under that new name.

For example, *ROOMFILE.TDS* becomes *ROOMF001.TDS* or *ROOMF001.TDS* becomes *ROOMF002.TDS* when *F3* is pressed.

The data for each measurement is stored under an eight-character file name. *Sound Lab TDS* will not accept more than eight characters for the file name, and it automatically adds a two- or three-character suffix (called an extension) to indicate the type of data.

The file name extensions are:

Time and distance	.ETC
Frequency	.TDS
STI and RASTI	.STI
Noise	.NC
TDS config. file	.TCF
Startup config.	.SET
Go to menu	.MNU
3D	.3D
Individual 3D	.###

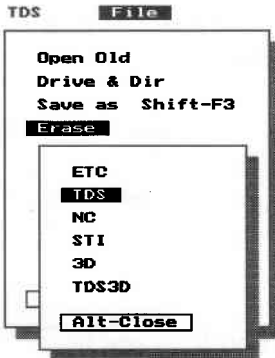


Illustration 4-6
Erase sub-menu

Erase - The command *Erase* allows the removal of data files chosen from a list box. The *Erase* sub-menu follows the same path navigation routine as described in *Open Old*. Before *Sound Lab TDS* erases the file, an information box will ask you to confirm your choice and choose *Yes* or *No*.

Configuration - Use *Configuration* to load, save or erase global setup data. Configuration files provide a convenient way to recall often-used parameters. Configuration files end in *.TCF* for TDS files. All configuration files must be located in the same directory as the *Sound Lab TDS* program file.

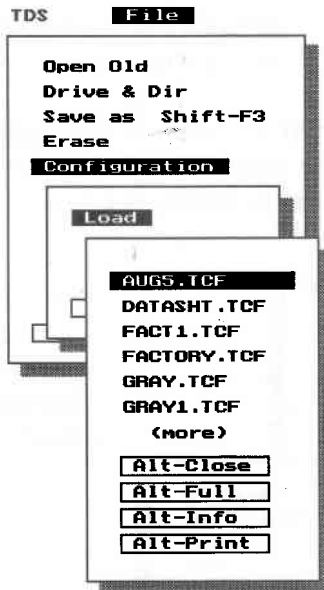


Illustration 4-7
Configuration sub-menus

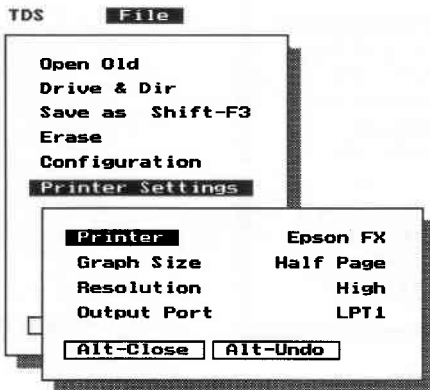
Printer settings - Select from the choices in the menu. Select a printer before choosing *Print*.

Graph Size — The half-page graph size option prints all data on one page. The full page option prints the graph on one page; test parameters and setup data are printed on a second page.

Resolution — The medium and high resolution options determine print quality. High resolution yields a better printout but takes longer to print than medium resolution.

Output Port — Select the port that matches the printer connection to your computer.

Print - *Sound Lab TDS* prints the current data screen (not including overlays) to the printer using settings made in *Printer Settings*.



*Illustration 4-8
Printer Settings sub-menu*

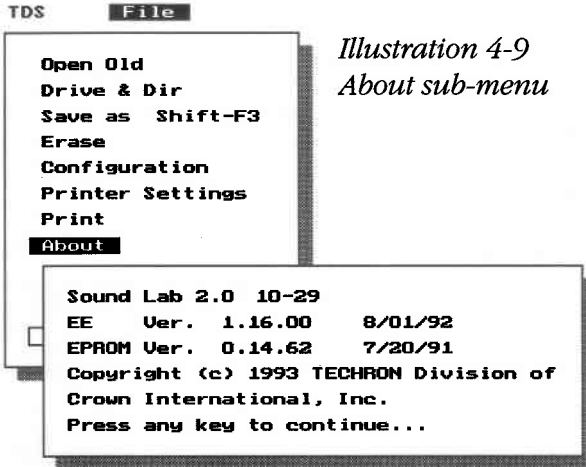
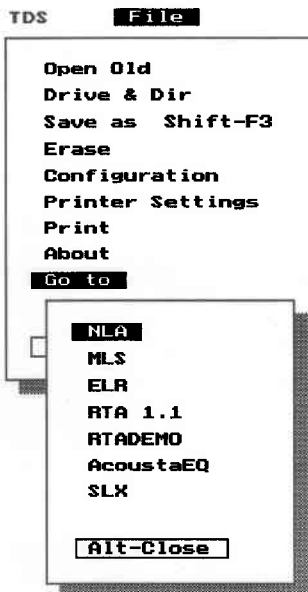


Illustration 4-9
About sub-menu

About - Displays the current version and date of the software and the firmware inside the TEF analyzer.

Note: The *About* command can be used as a diagnostic tool to check the connection between the PC and the TEF analyzer. If the PC is not communicating with the analyzer, a message will appear:

"The TEF is not ready. ROM data is not available."



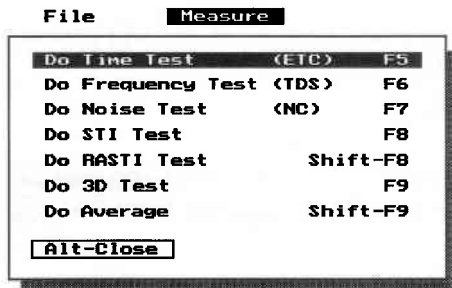
Go to—*Sound Lab* is composed of several modules that make up a total measurement package. *TDS* is a module within the *Sound Lab* family. When you start *Sound Lab TDS* for the first time, the TDS module is loaded. You choose other modules with the *Go to* command under the *File* menu. As you install new modules, they will appear in the *Go to* sub-menu.

Quit - returns you to DOS. After you select *Quit*, a sub-menu will appear and ask you, *Do you really want to quit?* Choose *Yes* or *No*.

Illustration 4-10
Go to sub-menu

Measure menu—Starting the test sequence

The commands in the *Measure* menu start the *Sound Lab TDS* test sequence. *Sound Lab* will perform the measurement with the settings made under the *Input* and *Parameters* menus. It will then display the results on the screen with the settings made under the *Display* menu.



The sub-menus under the *Measure* menu are:

Do Time Test - sends a sweep of selected frequencies into a system or device and calculates and displays the amplitude of energy received over a specified time period. The Time Response test is useful to pinpoint the arrival times of energy at a microphone. RT₆₀, %AL_{CONS}, and Early-to-Direct/

Early Reverberation information is derived from the Time Response.

Illustration 4-11
TDS Measure menu

Do Frequency Test - sends a sweep of selected frequencies into a system or device and calculates and displays an amplitude versus frequency plot. The frequency response is used to find the true response of a loudspeaker, system or device in the frequency domain.

Do Noise Test - gathers noise data over a specified period of time in eight octave bands and displays the results on a noise criteria(NC) graph showing octave band sound pressure levels versus frequency in eight octave bands. The lowest NC contour not exceeded by the curve becomes the NC value.

Do STI Test (*Speech Transmission Index*)—

Do RASTI Test (*RApid Speech Transmission Index*)—

performs STI (and the faster RASTI) tests by running several time response tests plus an NC measurement, and then processing the data to yield an overall speech intelligibility prediction. The resulting data is displayed on the screen in a graph or table.

- STI makes modulation transfer function (MTF) measurements in the seven octaves from 125 Hz to 8 kHz.
- RASTI is a shortened version of the STI procedure, measuring the MTF at the 500 Hz and 2 kHz octaves.

When you choose *Do STI Test*, *Do RASTI Test* and *Do 3D Test*, a *Save as* sub-menu appears because you must name the test before it begins.

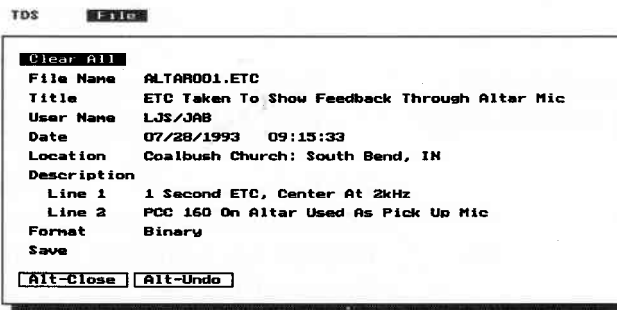


Illustration 4-12
Save as sub-menu

You can set the parameters to do a noiseless test and edit in noise later from a stored NC file or the NC table in the Display menu.



Do 3D Test—performs from 2 to 100 individual TDS measurements, each offset in time by a constant amount. When all of the individual sweeps are displayed on the screen, they form a three-dimensional display that plots the changes in magnitude/frequency over time.

Do Average - repeats from 2-999 tests and displays a the vector or magnitude average of Time or Frequency Response measurements. Signal averaging is used to improve signal-to-noise ratios under adverse conditions such as noise.

Note: You can cause an *Auto Repeat* mode for any Time, Frequency or Noise Test by pressing the *Function key F10*. The test will repeat until you press *Esc* to cancel it.

A logical test sequence in a room for time and frequency measurements

1. Make a coarse Time Response Test (one second span) to reveal energy arrival times at the microphone.
2. Refine the time resolution (0.1 to 0.5 second time span) if the first reading is too coarse for clarity.
3. Turn the cursor *On* and pinpoint the exact time of the first energy arrival at the microphone.
4. Do a Frequency Response Test using a receive delay which equals the arrival time from the Time response.

Parameters menu— setting test parameters

The commands in the *Parameters* menu determine the settings the TEF uses to make a test. After the test, changing any parameters removes test results from memory and erases the screen when the menu closes. Use *Alt-Undo* to retain parameters and not lose data.

Parameters

Time Response (ETC)

Frequency Response (TDS)

Noise Test (NC)

STI/RASTI (STI)

3D Test (3D)

Averaging

Output

Pink Noise	Off
White Noise	Off
Sine Wave Tone	Off
Frequency	0.0 Hz

Alt-Close

Several of the *Sound Lab* test parameters for the Time Response (ETC) test are interdependent; entry of one parameter may change the value of another. To keep the complete parameter set valid, *SoundLab* forces an entered parameter to its nearest valid value or issues a message to help you set valid parameters.

Illustration 4-13
The parameters menu

If you receive a message, adjusting another parameter may allow you to enter a parameter in the range you desire.

SoundLab rounds all entered parameters to the nearest valid value. For example, if you enter a time span of 100 seconds, the actual value entered by *Sound Lab* would be 998.1652 seconds. This slight change results from the digital implementation of the sweep. *Sound Lab* allows only certain sweep rates to be selected.

Time Response (ETC) Parameters

The interdependent ETC parameters are: *Start frequency, stop frequency, center frequency, frequency span, sweep time, receive delay, time span, number of samples and window.* Entry of one of these parameters may change the value of other ETC parameters to keep the complete parameter set valid. *Table 4-1* shows the ETC interdependent parameters relationships.

Parameters

Time Response (ETC)

Start Frequency	100.0 Hz
Stop Frequency	10000.0 Hz
Center Frequency	5050.0 Hz
Frequency Span	9900.0 Hz
Sweep Time	1.0 s
Receive Delay	0.0000 ns
Time Span	43.9596 ns
Number of Samples	1024
Window	Hanning

Illustration 4-14

Parameters for the Time Response (ETC) test.

Start Frequency (ETC)—the starting frequency of the sweep.

Stop Frequency (ETC)—the ending frequency of the sweep.

Center Frequency (ETC)—the frequency halfway between the start and stop of the sweep. This is entered automatically when you set start and stop frequencies.



Frequency Span (ETC)—range of frequencies, start to stop, over which the TEF sweeps.

Sweep Time (ETC)—the duration of a TEF sweep.

Receive Delay (ETC)—the difference in time between the start of the sweep and when the analyzer starts listening for the return signal.

Time span (ETC)—the time during which we listen for the effects of the signal on the room or system. It is shown in Time Response (ETC) measurements on the x axis on the screen.

Number of samples (ETC)—the number of points, during a sweep, at which data will be recorded.

Window (ETC)—the user may select a Blackman, Hamming, Hanning, Kaiser or rectangular window for use in processing the data.

Note: *Sound Lab* will not allow you to change to a frequency span or time span that would require a start frequency below 100 Hz. Instead, *Sound Lab* will make the following settings:

- Set start frequency to 100 Hz
- Calculate and set a new stop frequency to maintain the same frequency span.

Note: A start frequency below 100 Hz may be entered manually.



Guidelines for ETC measurement parameters:

- For the initial room measurement, calculate how long it will take sound to travel the longest dimension of the room. Set the time span to be at least 10 times longer than the travel time.
- Set sweep time to be 3 to 4 times longer than the time span. Sweep time effects the quality of the measurement—the longer the sweep time, the greater the noise immunity.
- Select a center frequency that matches the center of the frequency range of interest. *Sound Lab* will calculate start and stop frequencies to maintain the selected time span. Stop frequency must be greater than start frequency by 14 Hz.
- Frequency span and number of samples change the time span. Time span is calculated from the following formula:

$$\begin{array}{l} \text{Time span} = \\ \text{in ms} \end{array} \quad \frac{1000 \times 0.85 \times \text{number of samples}}{2 \times (\text{stop freq.} - \text{start freq.})}$$

- Use the *Hamming* window function for performing acoustic measurements and the *Rectangular* window for performing electronic measurements.

ETC parameters		Resulting changes							
		Start frequency	Stop frequency	Center frequency	Frequency span	Sweep time	Receive delay	Time span	Number of samples
Change entered ↓	Start frequency	U	NC	C	C	NC	NC	C	NC
	Stop frequency	NC	U	C	C	NC	NC	C	NC
	Center frequency	C	C	U	NC	NC	NC	NC	NC
	Frequency span	C	C	NC	U	NC	NC	C	NC
	Sweep time	NC	NC	NC	NC	U	NC	NC	NC
	Receive delay	NC	NC	NC	NC	NC	U	NC	NC
	Time span	C	C	NC	C	NC	NC	U	NC
	Number of samples	NC	NC	NC	NC	NC	NC	C	U

Table 4-1 shows the interdependent parameter relationships for the Time Response test.

TDS parameters—Frequency Response

The interdependent parameters for Frequency Response (TDS) tests are: *start frequency*, *stop frequency*, *sweep time*, *sweep rate*, *resolutions*, *receive delay*, *bandwidth*, and *number of samples*. Entry of one of these TDS parameters may change the value of other TDS parameters to keep the complete parameter set valid. Table 4-2 shows the TDS dependent parameter relationships.

Parameters	
Time Response (ETC)	
Frequency Response (TDS)	
Start Frequency	100.0 Hz
Stop Frequency	10000.0 Hz
Sweep Time	1.0 s
Sweep Rate	10312.5 Hz/s
Resolution:	
Frequency	101.6 Hz
Distance	11.1 ft
Time	9.8473 ns
Best Frequency Resolution	On
Receive Delay	0.6125 ns
Bandwidth	101.6 Hz
Number of Samples	512
Alt-Close Alt-Undo	

Illustration 4-15

The parameters for the Frequency Response test

Start Frequency (TDS)—the starting frequency of the sweep.

Stop Frequency (TDS)—the ending frequency of the sweep.

Sweep Time (TDS)—the duration, in seconds, of a TEF sweep.

Sweep rate (TDS)—the rate in Hz/second of a TEF sweep.



Resolution (TDS)—the smallest increment that can be correctly discerned in a parameter you have chosen.

Frequency—the smallest increment of frequency that you will be able to resolve or see correctly.

$$\text{If Bandwidth} \leq \sqrt{\text{Sweep rate}}, \text{ then}$$

$$\text{Frequency resolution} = \frac{\text{Sweep rate}}{\text{Bandwidth}}$$

TDS

$$\text{If Bandwidth} > \sqrt{\text{Sweep rate}}, \text{ then}$$

$$\text{Frequency resolution} = \text{Bandwidth}$$

*Conventional
swept spectral
analysis*

Distance—the smallest interval in length you will be able to resolve or see correctly.

Time—the smallest increment of time that you will be able to resolve or see.

$$\text{Time resolution} = \frac{\text{Bandwidth}}{\text{Sweep rate}}$$

Best Frequency Resolution (TDS)—When *Best Frequency Resolution* is selected, *Sound Lab* automatically calculates the optimum bandwidth to produce the best frequency resolution (poorest time resolution) for the sweep time that you have chosen. Optimum bandwidth is the square root of the sweep rate (sweep rate is calculated from sweep time and frequency span).



Receive Delay (TDS)—the difference in time between the start of the sweep and when the analyzer starts listening for the signal to return.

Bandwidth (TDS)—The value entered here determines the size of sweeping filter, i.e., what the filter can see. See *Appendix A, How the TEF works.*

Number of samples (TDS)—the number of points, during a sweep, at which data will be recorded.

TDS Parameters		Resulting change →									
		Start frequency	Stop frequency	Sweep time	Receive delay	Frequency resolution	Distance resolution	Time resolution	Sweep rate	Bandwidth	# Samples
Change entered ↓ C= Change U= Update NC= No change	Start frequency	U	NC	NC	NC	NC	NC	C	NC	NC	NC
	Stop frequency	NC	U	NC	NC	NC	NC	C	NC	NC	NC
	Sweep time	NC	NC	U	NC	NC	NC	C	NC	NC	NC
	Receive delay	NC	NC	NC	U	NC	NC	NC	NC	NC	NC
	Frequency resolution	NC	NC	NC	NC	U	C	C	NC	C	NC
	Distance resolution	NC	NC	NC	NC	C	U	C	NC	C	NC
	Time resolution	NC	NC	NC	NC	C	C	U	NC	C	NC
	Sweep rate	NC	NC	C	NC	NC	NC	U	NC	NC	NC
	Bandwidth	NC	NC	NC	NC	C	C	C	NC	U	NC
	# Samples	NC	NC	NC	NC	NC	NC	NC	NC	NC	U

Table 4-2 Interdependent parameter relationships for the Frequency Response (TDS) test, Best Frequency Resolution OFF.



Guidelines for TDS measurement parameters:

- Use an ETC measurement to determine the exact time of arrival of the sound you want to analyze. Set the receive delay to equal the time of arrival.
- Set sweep time to be as short as possible to achieve the desired results. See *Appendix A, How TEF works*.
- Sweep time effects the quality of the measurement—the longer the sweep time, the greater the noise immunity.
- Select start and stop frequencies that cover the frequency range of interest. Stop frequency must be greater than start frequency by 1 Hz.
- Start frequency must be greater than or equal to the bandwidth for a valid test.
- When *Best Frequency Resolution* is *On*, *Sound Lab* calculates the optimum bandwidth and all resolutions (see the next section.)
- Bandwidth must be greater than or equal to 2 Hz and less than or equal to 240 Hz.



Best frequency resolution and optimum bandwidth

When *Best Frequency Resolution* is *On*, *Sound Lab* automatically calculates the optimum bandwidth to produce the best frequency resolution (poorest time resolution) for the sweep time that you have chosen. Optimum bandwidth is the square root of the sweep rate (sweep rate is calculated from sweep time and frequency span).

When the bandwidth is larger than the square root of the sweep rate, the frequency resolution is equal to the bandwidth. Under this condition, the TEF performs similarly to conventional swept sine wave analysis and will properly measure the peak amplitudes of narrowband stationary signals (such as hum and noise); however, fine detail may be passed over in the frequency response. Also, because time resolution decreases, reflections may be included in the measurement of the direct sound.

C= Change
U= Update
NC= No change

**Change
entered**



**TDS
Parameters**

**Resulting
change** →

	Start frequency	Stop frequency	Sweep time	Receive delay	Frequency resolution	Distance resolution	Time resolution	Sweep rate	Bandwidth	# Samples
Start frequency	U	NC	NC	NC	C	C	C	C	C	NC
Stop frequency	NC	U	NC	NC	C	C	C	C	C	NC
Sweep time	NC	NC	U	NC	C	C	C	C	C	NC
Receive delay	NC	NC	NC	U	NC	NC	NC	NC	NC	NC
Frequency resolution	Calculated by Sound Lab									
Distance resolution										
Time resolution										
Sweep rate	NC	NC	C	NC	C	C	C	U	C	NC
Bandwidth	Calculated by Sound Lab									
# Samples	NC	NC	NC	NC	NC	NC	NC	NC	NC	U

*Table 4-3
Interdependent parameter relationships for
Frequency Response (TDS) tests, Best Frequency
Resolution On.*

Parameters—Noise Test (NC)

To perform a noise test, you set the *Integration Time* and specify if you are using the *Line Output* of the TEF.

Parameters

Time Response (ETC)
Frequency Response (TDS)
Noise Test (NC)

Integration Time 4.0 s
Line Output Off

Alt-Close Alt-Undo

White Noise Off
Sine Wave Tone Off
Frequency 1000.0 Hz

Alt-Close

Illustration 4-16 The parameters for the Noise Test

Integration Time—Enter a value here (1-9999 sec) for the length of time (in seconds) that the data is to be averaged on each individual octave.

Line Output—*On* or *Off*. When turned on, the *Line Output* allows you to monitor the octave-band filtered noise at the output of the TEF through headphones, an amplifier or a tape recorder.



Guidelines for setting NC parameters.

Input levels have to be sufficient to yield valid results. If the levels are set too low, the resulting graphs will show abnormally low noise curves, especially at the higher frequencies.

- If you are using the built-in TEF preamps, adjust the gain in the preamp until you get an overload, then lower it one step to get the proper level.
- If you are using a sound level meter and the line level input, the output of the sound level meter should be in the range of 1 millivolt to 1 volt to perform valid NC measurements.
- Due to the inherent noise of the computer and TEF, your measurement setup should be in a different room from where the NC is being measured.

Parameters—STI/RASTI (STI)

When you choose *Parameters...STI/RASTI*, a sub-menu will appear and allow you to turn on *Measure Noise* and/or *Source Calibration*.

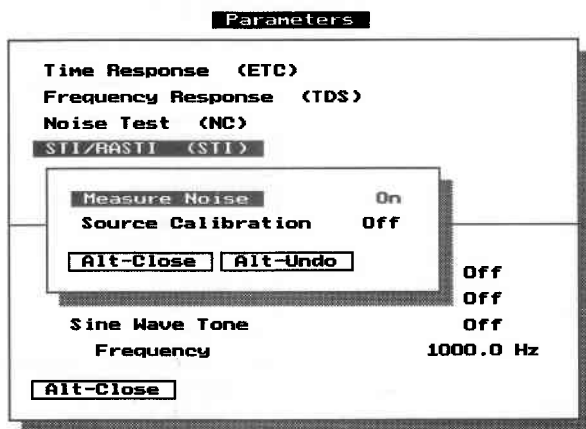


Illustration 4-17

The STI/RASTI parameters menu

Measure Noise—This choice is an *On* or *Off* toggle. *Sound Lab* provides the option to measure either with or without background noise. Choose *Off* if you do not want the room noise included in the measurement. If you measure without noise, it can be entered later by choosing *Add Noise to STI/RASTI* in the *Display* menu.

Source Calibration—This choice is an *On* or *Off* toggle. For correct speech intelligibility measurements, the acoustic level of the source loudspeaker must be calibrated to simulate the actual level of human speech. See *Section 4, Measure menu* for the complete source calibration procedure.

Parameters—3D Test (3D)

The *Parameters* menu for 3D is similar to the parameters for the Frequency Response test. The interactions among the following values of *Start receive delay*, *End Receive Delay*, *No. of Curves* and *Receive Delay Step* are shown in Table 4-4.

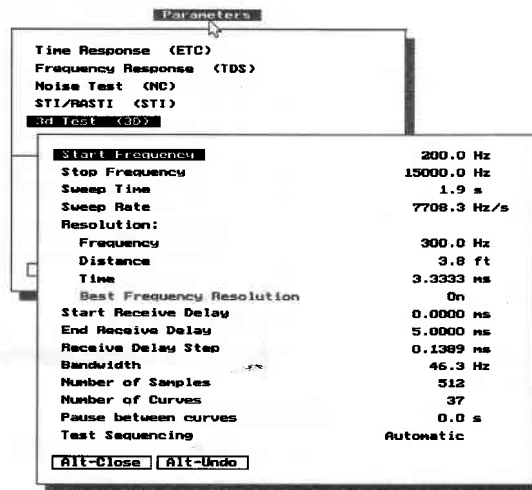


Illustration 4-18
The 3D parameters menu

Start Frequency (3D)—the starting frequency of the sweep.

Stop Frequency (3D)—the ending frequency of the sweep.

Sweep Time (3D)—the duration of a TEF sweep.

Sweep rate (3D)—the rate in Hz/second of a TEF sweep.

Resolution (3D)—the smallest increment that can be correctly discerned in a parameter you have chosen.

Frequency—the smallest increment of frequency that can be correctly discerned in the frequency domain.

Distance—the smallest interval in length you will be able to resolve or see correctly.

Time—the smallest increment of time you are able to see or correctly resolve in the time domain.

Best Frequency Resolution (3D)—When *Best Frequency Resolution* is selected, *Sound Lab* automatically calculates the optimum bandwidth to produce the best frequency resolution (poorest time resolution) for the sweep time that you have chosen. Optimum bandwidth is the square root of the sweep rate (sweep rate is calculated from sweep time and frequency span).

Start and End Receive Delay (3D)—These two parameters determine the time span between the first and last measurements. The software will allow the values to be equal, making this measurement useful for collecting polar data.

Start Receive delay (3D)—The receive delay of the first curve measured is stored and displayed as curve 001.

The *Start Receive Delay* can be manually entered into the menu, You can also use the cursors to enter the value directly into the parameters by using the cursor buttons that appear below an ETC display when the cursor is turned on. Position the ETC cursor at the desired data point on the display, and press the *S* key. An information box appears showing the *3D Start Delay* and *3D End Delay* values as determined by the cursor position.

End Receive delay (3D)—The receive delay of the last curve measured. When displayed with the cursor, this curve number is equal to the value at *No. of Curves*.

The *End Receive Delay* can be manually entered into the menu, You can also use the cursors to enter the value directly into the parameters. When using the cursor buttons in an ETC display, four buttons will appear below the display. Position the ETC cursor at the desired data point and press the *S* key. An information box appears showing the *3D Start Delays* set before and the current *3D End Delay* values.

Receive Delay Step (3D)—The value entered here sets the receive delay change between the individual curves.

Bandwidth (3D)—The value entered here determines the size of the sweeping filter, i.e., what the filter can see. See *Appendix A, How the TEF works*.



Number of samples (3D)—The value entered here determines the number of points at which data will be recorded on the TEF during a sweep.

Number of Curves (3D)—*Sound Lab* will make from 2 to 100 curves in one measurement set.

Pause Between Curves (3D)—A value entered here determines the number of seconds the TEF pauses before starting the next sweep.

This feature is useful when you need to change something in the physical test arrangement between sweeps. For example, if you are collecting polar data and need to rotate a loudspeaker between sweeps, enter a value in *Pause Between Curves* that allows enough time to reposition the speaker and step out of the measurement sound field.

Test Sequencing (3D)—When you choose *Test Sequencing*, a sub-menu allows you to select one of four ways to start the sweep. The choices are *Automatic*, *Keyboard*, *DC Shift Trigger*, and *Pulse Trigger*.

Automatic—When you choose *Automatic*, *Sound Lab* automatically makes each sweep with the pause between them the number of seconds as set in *Parameters...3D Test...Pause between curves*.

Keyboard—When you choose *Keyboard*, the software waits for you to press a key before making the next sweep.

DC Shift Trigger—This option is available to work with a remote push button accessory connected to the TEF through the external trigger connector. When *DC Shift Trigger* is selected, the software begins the measurement when signaled by a push button. (See schematic in analyzer user manual)

Pulse trigger—This option was designed specifically to be used with an accessory turntable* connected to the TEF through the external trigger connector. For specific turntable details, refer to the instructions from the manufacturer.

*Outline® Outline snc, Via Leonardo da Vinci, 56
25020 Flero (BS), Italy, 030/3581341, FAX 030/
3580431.



3D Curve Spacing

C= Change
U= Update
NC= No change

**Change
entered**



**Resulting
change** →

	<div>Start Receive Delay</div> <div>End Receive Delay</div> <div>Receive Delay Step</div> <div>No. of curves</div>			
Start Receive Delay	U	NC	NC	NC
End Receive Delay	NC	U	C	NC
Receive Delay Step	C	C	U	C
No. of curves	NC	NC	N C	U

Table 4-4

This table shows 3D curve spacing interaction. For interaction among other parameters, see previous tables.

Parameters for Averaging

Sound Lab allows you to perform a vector or magnitude average of two to 999 frequency or time measurements.

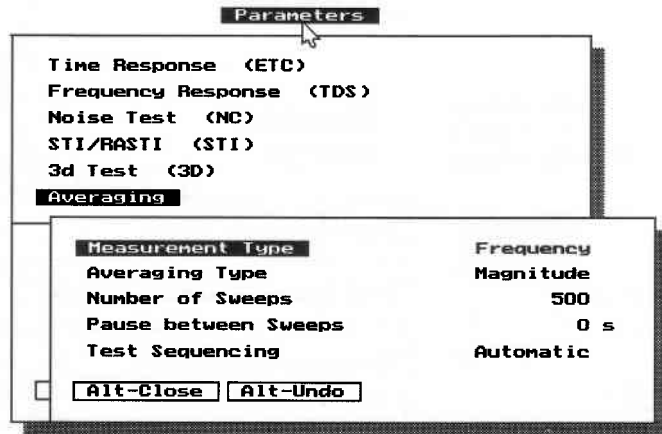


Illustration 4-19

Parameters for Averaging menu and sub-menu.

Measurement type (Avg)—Toggles between *Frequency* and *Time*. Choose the measurement type you wish to average.

Averaging Type (Avg)—Toggles between *Vector* or *Magnitude*. Choose the averaging type you prefer.

Number of Sweeps (Avg)—*Sound Lab* will make from 2 to 999 sweeps in the averaging process.

Pause Between Sweeps (Avg)—A value entered here determines the number of seconds the TEF waits between sweeps.



Test Sequencing (Avg)—When you choose *Test Sequencing*, a sub-menu allows you to select one of four ways to start the sweep. The choices are *Automatic*, *Keyboard*, *DC Shift Trigger*, and *Pulse Trigger*

Automatic—When you choose *Automatic*, *Sound Lab* makes each sweep with the pause between them the number of seconds set in *Parameters...3D Test...Pause between curves*.

Keyboard—When you choose *Keyboard*, the software waits for you to press a key before making the next sweep.

DC Shift Trigger—This option is available to work with a remote push button accessory connected to the TEF through the external trigger connector. When *DC Shift Trigger* is selected, the software begins the measurement when signaled by a push button. (See schematic in analyzer user manual)

Pulse trigger—This option was designed specifically to be used with an accessory turntable connected to the TEF through the external trigger connector. For specific turntable details, refer to the instructions from the manufacturer.



Guidelines for averaging

Vector averaging

With vector averaging, the complex data (real and imaginary) is averaged for corresponding points in the multiple sweeps. Use vector averaging for improved signal to noise ratio. This improvement is somewhat reduced if the signal moves in time due to air currents.

Magnitude averaging

With magnitude averaging, the magnitude of the complex data (real and imaginary) is averaged for corresponding points in the multiple sweeps. Magnitude averaging is useful for finding an average sound level for a region of the room when the microphone placed in several locations and then moved between sweeps.

Output

The options under *Output* enable you to command the TEF to generate *Pink* or *White Noise*, or a *Sine Wave Tone* up to 24000 Hz.

Pink Noise is activated by choosing the *Pink Noise* option. White Noise is activated by choosing the *White Noise* option.

To specify a frequency for the *Sine Wave Tone*, choose *Frequency* and enter a value for Hz. When you choose *Sine Wave Tone*, the tone will be heard. The *Enter* key will toggle on and off any of the selected options.

Parameters

Time Response (ETC)

Frequency Response (TDS)

Noise Test (NC)

STI/RASTI (STI)

3D Test (3D)

Averaging

Output

Pink Noise	Off
White Noise	Off
Sine Wave Tone	Off
Frequency	0.0 Hz

Alt-Close

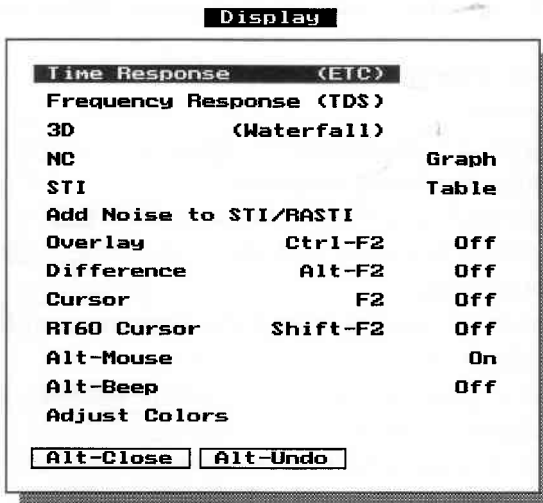
Illustration 4-20

The Output section of the Parameters menu.

Display pull-down menu

In the *Display* menu, you control the way data is displayed on the screen. You set the parameters for each display, but you do not need to do this every time you make a measurement. Instead, you can change the display after the measurement is done. Also in the Display menu, you can:

- Turn On and Off display modes such as Cursor, Difference and Overlay.
- Turn On and Off the mouse and the warning beep
- Enter noise data into noiseless STI and RASTI data, or add new noise into measurements already taken with noise.
- Change the display colors



Display options

The options for the display of data from measurements are arranged under the *Displays* sub-menu according to measurement type such as Time and Frequency Response, 3D, NC, and STI/RASTI.

Illustration 4-21
Display menu

Display options by measurement

Time Response—Displays time response or the output of a system vs. time.

There are two types of time response displays which you can select under the menu item called *Time Response*.

Magnitude: Shows energy vs. time.

Heyser Spiral: Shows the complete system response in one view—a three-dimensional curve in which the three projections are the polar, real, and imaginary planes.

Frequency Response—Displays the frequency response.

There are five types of frequency response displays which you can select under the item called *Frequency Response*.

Phase: Phase vs. frequency, or phase response.

Magnitude: Magnitude vs. frequency, or the familiar frequency response

Magnitude and Phase: Displays both the frequency response and phase response on the same screen.

Nyquist: A rotating vector showing magnitude vs. phase angle on a polar plot.

Heyser Spiral: Shows the complete system response in one view—a three-dimensional curve in which the three projections are the polar, real, and imaginary planes.

3D (Waterfall)—Displays frequency response vs. time in successive time slices.

NC (Noise Criteria)—Noise contour graph showing octave band sound pressure levels vs. frequency in eight octave bands.

STI (Speech Transmission Index)—A measure of the intelligibility of a sound system.

Display options—Time Response...magnitude

When you select *Time Response...Display type* a submenu allows you to choose two different ways to view the measurement data—*Magnitude* or *Heyser Spiral*. The parameters which can be set for the *Time Response—Magnitude* display are:

Top of Scale
Bottom of Scale
Auto Scaling

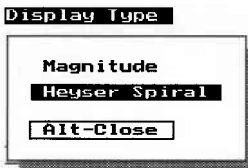


Illustration 4-22
Time response (ETC)
sub-menu

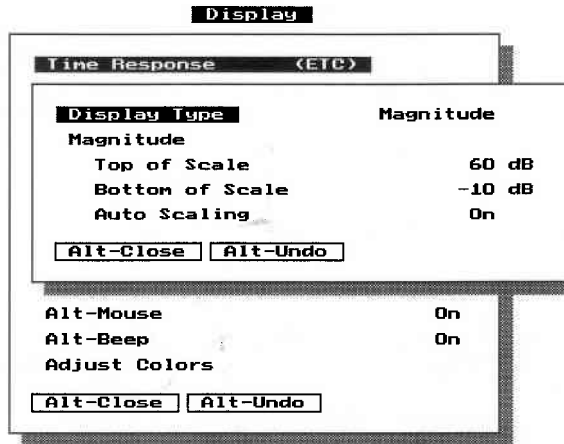


Illustration 4-23
Display...Time Response submenu

Top of Scale—The *Top of Scale* value (in dB) determines the value that will be displayed at the top of the vertical scale. This data entry value will be rounded to the nearest 10 dB increment.



Bottom of Scale—The *Bottom of Scale* value (in dB) determines the value that will be displayed at the bottom of the vertical scale. This data entry value will be rounded to the nearest 10 dB increment.

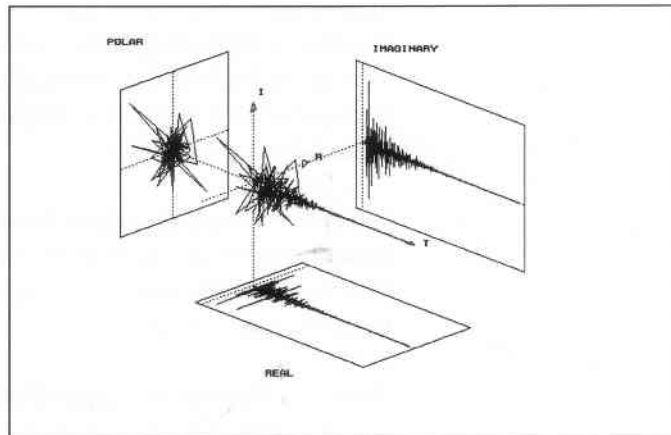
Auto Scaling—Toggle *Auto Scaling* on or off. If *Auto Scaling* is *On*, the software will automatically scale the data to display the full magnitude range of the measurement. The highest data value is placed in the top 10 dB of the graph, and the scale annotation is adjusted accordingly. The new *Top of Scale* and *Bottom of Scales* will be updated when the data is redisplayed. Manually entering a value for *Top of Scale* or *Bottom of Scale* automatically toggles *Auto Scaling Off*.

We recommend that *Auto Scaling* be left on for most measurement tasks. If *Auto Scaling* is off and all the data is out of the selected range, you will not see any data on the screen.

Auto Scaling can be toggled off if you don't want the graph's *Top of Scale* value to change with data level from test to test. This is a requirement if you are going to use *Overlay* to create a family of curves on the screen resulting from several tests. See additional comments in *Overlay* command description.

Display options— Time Response...Heyser Spiral

The *Heyser Spiral* display shows the complete system response in one view—a three-dimensional curve in which the three projections are the polar, real, and imaginary planes. When the display is on the screen, you can press *F1* to open a sub-menu which gives key combinations for various manipulations of the display.



*Illustration 4-24
The Heyser Spiral
display of a Time
Response test.*

*Illustration 4-25
The Help menu,
appears when you
press F1.*

Display	
Rotate plot left or right	Left or Right arrow
Tilt plot up or down	Up or Down arrow
Move plot up or down	PageUp or PageDown
Move plot left or right	< or >
Increase rotation and move steps	Alt-I
Decrease rotation and move steps	Alt-D
Expand time and frequency axis	Alt-X
Compress time and frequency axis	Alt-C
Return to original position	Home
press any key to continue	

Display options—Frequency Response

When you select *Display...Frequency Response...Display Type*, you may choose one of five display types:

- Phase (vs. frequency)**
- Magnitude (vs. frequency)**
- Magnitude and Phase**
- Nyquist**
- Heyser Spiral**

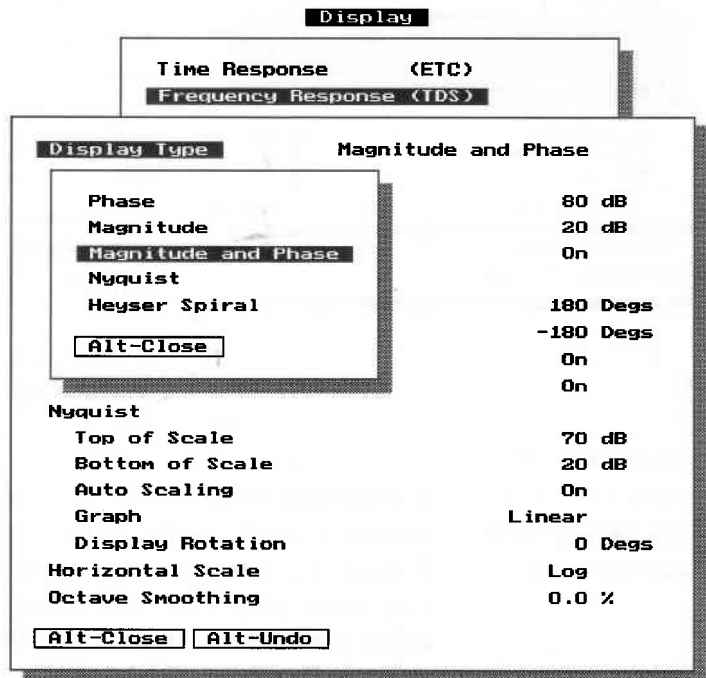
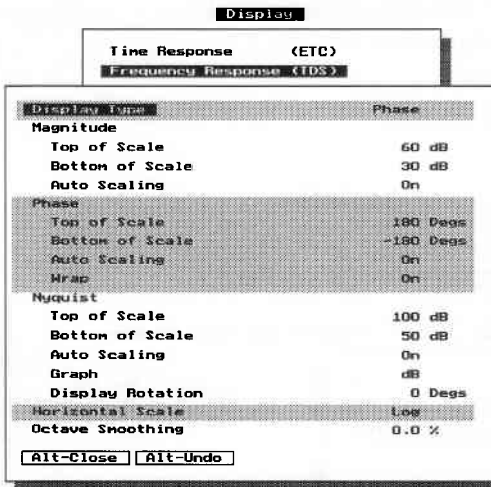


Illustration 4-26
Frequency Response Display sub-menu

Display options—

Frequency Response...Phase

The parameters that affect the *Phase* display are *Top of Scale*, *Bottom of Scale*, *Auto Scaling*, *Wrap* and *Horizontal Scale* (shown in gray).



Wrap—When *Wrap* is toggled *On* the phase curve will wrap or start over at the top of the scale as frequency progresses. The top of the graph will be 180 degrees and the bottom will be -180 degrees. With *Wrap* off, the phase curve will be continuous, without any sudden 360-degree phase jumps or transitions.

If you edit the *Top of Scale* or *Bottom of Scale*, *Wrap* turns off.

Illustration 4-27
Parameters for the
Phase display of the
frequency response

Horizontal Scale—Choose a linear or logarithmic display to set the horizontal frequency display scale. Display the data on a *Log* scale to see a traditional frequency response; use *Linear* to display signal-delay problems such as comb filtering.

Display options—

Frequency Response...Magnitude

The parameters that affect the Frequency Response—Magnitude display are *Top of Scale*, *Bottom of Scale*, *Auto Scaling*, *Horizontal Scale*, and *Octave Smoothing*.

Display

Time Response (ETC)
Frequency Response (TDS)

Display Type	Magnitude
Magnitude	
Top of Scale	60 dB
Bottom of Scale	30 dB
Auto Scaling	On
Phase	
Top of Scale	180 Degs
Bottom of Scale	-180 Degs
Auto Scaling	On
Wrap	On
Nyquist	
Top of Scale	100 dB
Bottom of Scale	50 dB
Auto Scaling	On
Graph	dB
Display Rotation	0 Degs
Horizontal Scale	Log
Octave Smoothing	0.0 %

Alt-Close
Alt-Undo

Illustration 4-28

Parameters for Frequency response magnitude display

Octave smoothing—Enter a percentage value for desired amount of octave smoothing to be performed on the data to be displayed. The value may be entered as a fraction, but it will be displayed in the menu as a percent.

Display options—Frequency Response... Magnitude and Phase

The options that affect the *Frequency Response* —
Magnitude and *Phase* display are:

- **Magnitude**—*Top of Scale, Bottom of Scale, Auto Scaling, Octave smoothing*
- **Phase**—*Top of Scale, Bottom of Scale, Auto Scaling, Wrap, Horizontal Scale,*

See prior descriptions for *Magnitude* and *Phase* displays.

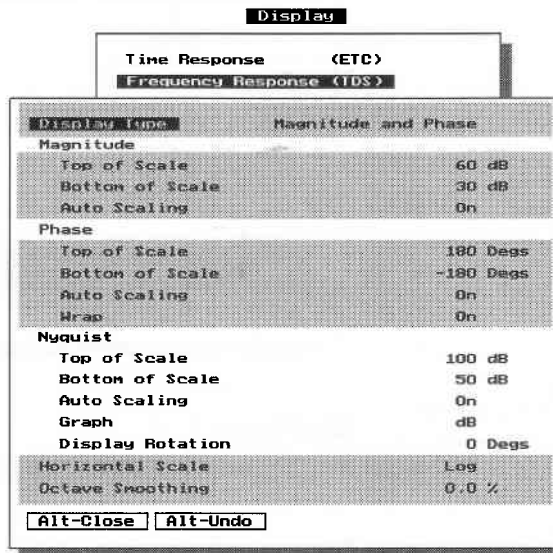


Illustration 4-29
Display parameters for Magnitude and Phase display

Display options— Frequency Response...Nyquist

The options for the *Nyquist* data display are *Top of Scale*, *Bottom of Scale*, *Auto Scaling Graph* and *Display Rotation*, and *Octave smoothing*.

Display	
Time Response (ETC)	
Frequency Response (TDS)	
Display Type	Nyquist
Magnitude	
Top of Scale	60 dB
Bottom of Scale	30 dB
Auto Scaling	On
Phase	
Top of Scale	180 Degs
Bottom of Scale	-180 Degs
Auto Scaling	On
Wrap	On
Nyquist	
Top of Scale	100 dB
Bottom of Scale	50 dB
Auto Scaling	On
Graph	dB
Display Rotation	0 Degs
Horizontal Scale	Log
Octave Smoothing	0.0 %
<div>Alt-Close</div> <div>Alt-Undo</div>	

Illustration 4-30
Nyquist display parameters

The Nyquist display plots magnitude versus phase angle. It can be thought of as the tip of a vector that is changing in both length and angle as the frequency sweeps. The length of the vector is proportional to the magnitude of the data, and the angle of the vector represents the phase of the data.

When the data cursor (*F2*) is used, a readout appears in the lower left corner of the display showing the frequency, phase, magnitude, and the real and imaginary values.

Graph—Toggle to choose *dB* or *Linear* data in the Nyquist display.

Display Rotation—Data entry for the degrees of counter clockwise rotation you want in the Nyquist display, starting from zero degrees.

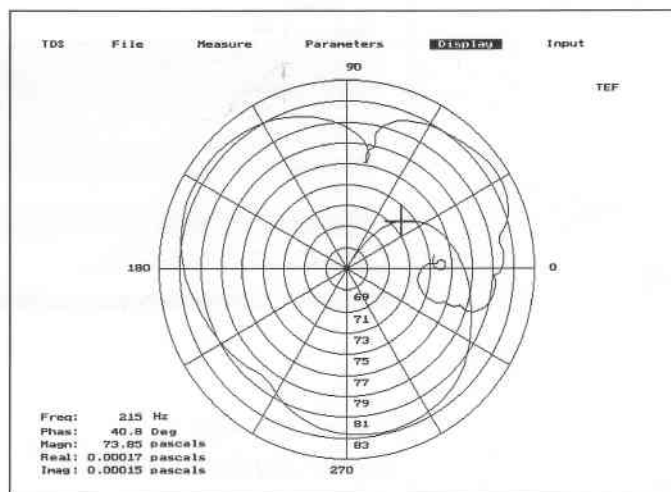


Illustration 4-31
The Nyquist display

Display options—

Frequency Response...Heyser Spiral

The amplitude of a complex waveform is the result of two factors called the real part and the imaginary part. In a plane progressive wave in the free field (i.e., before reflections occur) the real part is proportional to potential energy and the imaginary part is proportional to kinetic energy.

The amplitude is equal to:

$$\text{Amplitude} = \sqrt{(\text{real part})^2 + (\text{imag. part})^2}$$

and the magnitude is Magnitude =

$$\text{Magnitude} = 10 \log \sqrt{(\text{real part})^2 + (\text{imag. part})^2}$$

The phase response can be found by

$$\text{Phase} = \arctan \left(\frac{\text{imag. part}}{\text{real part}} \right)$$

The Heyser spiral reveals that amplitude and phase are simply two different viewpoints of the same event. Why measure phase?

- Phase is a more sensitive parameter to find the center frequency of filters.
- Phase reveals driver displacement, something amplitude can only do very coarsely. (i.e. inverse square law) whereas phase can show minute fractions of an inch.
- Phase is a clear detector of polarity.
- Phase reveals non-minimum phase behavior.

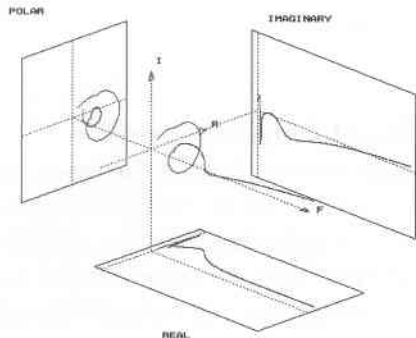


Illustration 4-32
Electrical impedance of a two way speaker.

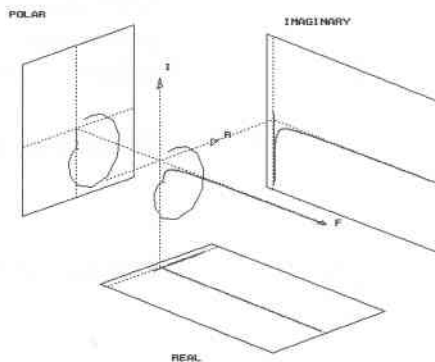


Illustration 4-33
Electrical impedance of a 20 microfarad capacitor

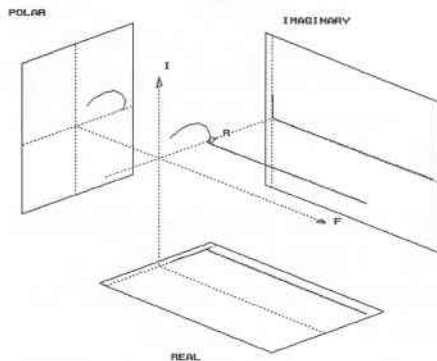


Illustration 4-34
Electrical impedance of an 8 ohm resistor

Display options—3D (Waterfall)

The 3D display shows frequency-response curves vs. time in successive time slices. Each curve is offset slightly from the previous one in the same display.

The parameters that can affect the 3D display are: *Perspective*, *Curve Order*, *Top of Scale*, *Bottom of Scale*, *Auto Scaling*, *Horizontal Scale*.

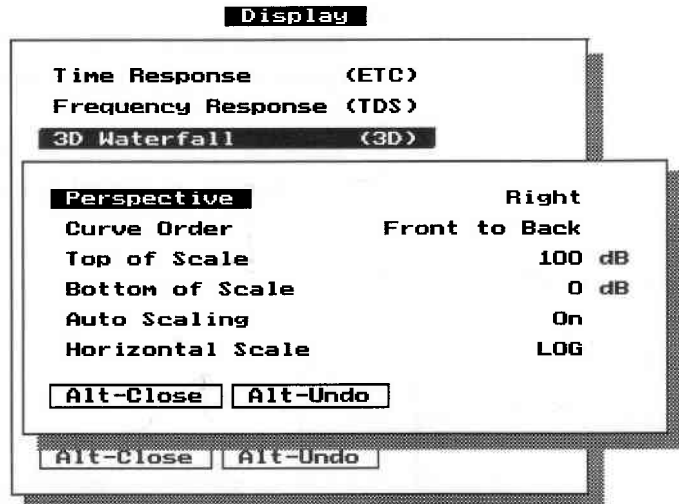


Illustration 4-35
The Display 3D Waterfall submenu

Perspective—Toggles between *Left* and *Right* views of the display. When viewed from the left perspective, the lower frequencies are nearest you, when viewed from the right perspective, the higher frequencies are nearest you.

Curve Order—Toggles between *Front to Back* and *Back to Front*, describing the placement on the graph of the first curve measured. The first curve is the one that is measured at the time set by *Start Receive Delay*.

Front to Back—The first curve measured (curve 001) is placed in the front of the graph.

Back to Front—The first curve measured (curve 001) is placed in the back of the graph.

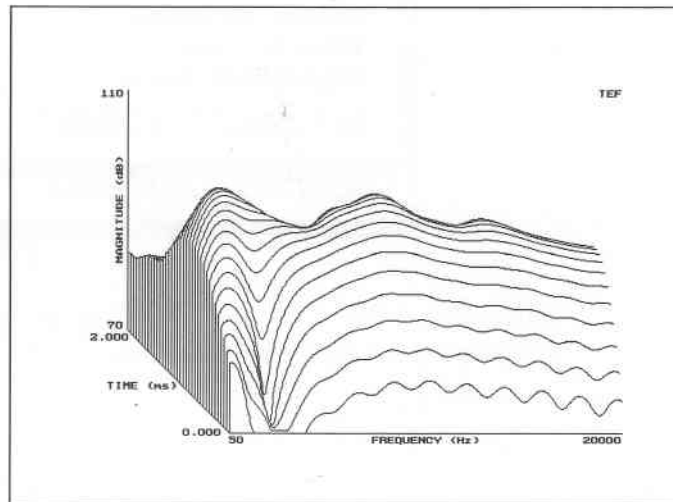


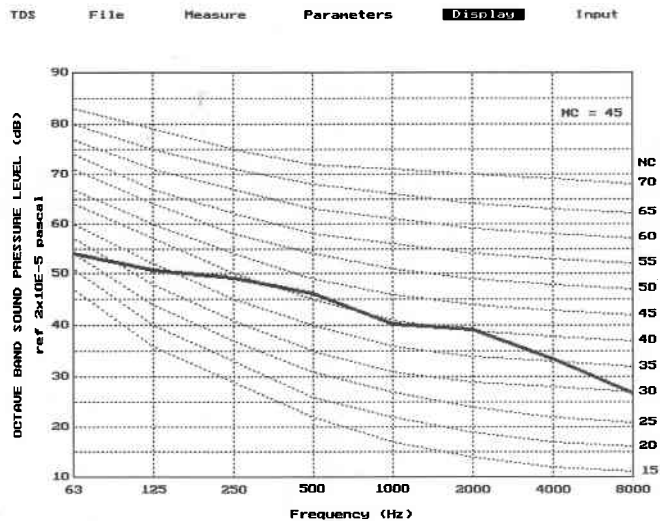
Illustration 4-36
The 3D Waterfall display

Display options—NC (Noise)

NC—Toggle this command to display the noise data as a Table or Graph.

Illustration 4-37
NC data displayed as a table
(right) and graph (below).

NOISE TABLE		TEF
Octave	Level	
63	54 dB	
125	51 dB	
250	49 dB	
500	46 dB	
1000	40 dB	
2000	39 dB	
4000	33 dB	
8000	27 dB	
NC = 45		



Display options—STI

STI—This command toggles to display STI measurements as a *Table* or *Graph*.

Illustration 4-38

Table displays the STI value for each band plus the overall STI value.

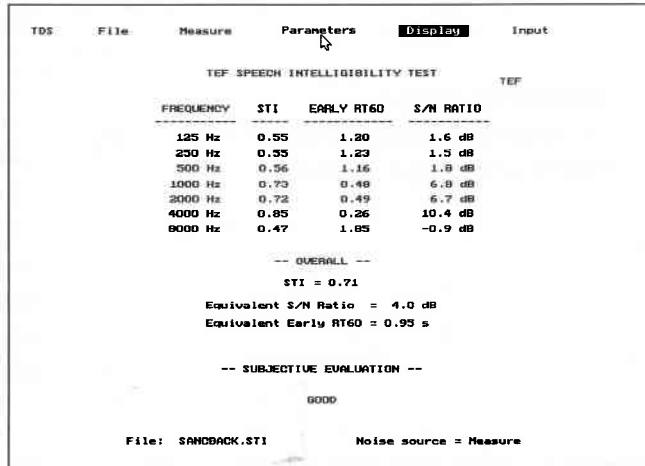
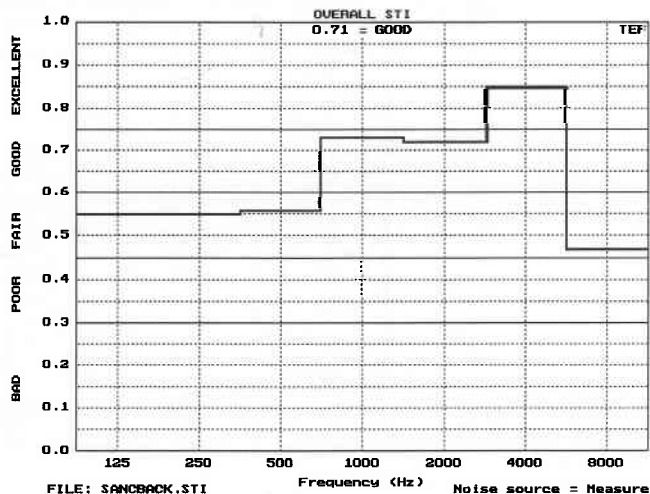


Illustration 4-39

Graph displays the STI values on a graph.



Add Noise to STI/RASTI

Add Noise from:
NC File
Table
Alt-Close

*Illustration 4-40
Add Noise To STI/
RASTI submenu*

Other Display options—

Add Noise to STI RASTI

The *Add Noise to STI/RASTI* option provides a way to process noise with noiseless STI or RASTI data. If you have no STI data on the screen, you will be given a message:

Obtain STI data before adding noise. Press any key to continue...

When you have STI data displayed on the screen, choosing *Add Noise to STI/RASTI* opens a sub-menu that allows you to choose noise values from an *NC file* or *Table*. If you choose *NC File*, a sub-menu opens showing available files from which to extract noise data. If you choose *Table*, a sub-menu will appear allowing you to enter noise values on octave band centers from the keyboard and command the software to *Recalculate* the data. The software will perform an FFT on the data with the noise included and redisplay the results.

*Illustration 4-41
Add noise options
under the Display
menu: (a) from a file
(b) from a table.*

TDS File

CFANOFF.NC
CFANON.NC
CFANONAS.NC
[...]

Alt-Close
Alt-Full
Alt-Info
Alt-Print

Add Noise to STI/RASTI

Noise Table

Octave	dB
a: 63	57 dB
b: 125	55 dB
c: 250	53 dB
d: 500	44 dB
e: 1000	28 dB
f: 2000	3 dB
g: 4000	-30 dB
h: 8000	20 dB

Recalculate

Alt-Close Alt-Undo

Other Display Options—Overlay

Sound Lab TDS allows you to overlay time, frequency or NC measurements. *Overlay* allows multiple curves of the same type to be placed on the screen at the same time. If *Overlay* is *On*, the screen is not erased between each measurement. Each new measurement curve is simply drawn over the curves of the previous tests. Data stored on disk may be overlaid by turning on *Overlay* and choosing *Open Old...* from the *File* menu.

Overlay only works for data gathered for common measurement types. For example, if a time response is on the screen, you cannot overlay a frequency response.

Any command that erases the screen will destroy any multiple curves built up on the display. For example, if *Overlay* is *On*, and a series of frequency response curves is displayed on the screen, performing a time response test will erase the screen.

The overlaid curves exist visually only on the screen. If the cursor is turned on when multiple curves are on the screen, the cursor will read only the data from the most recent test. *Auto scaling* is ignored when *Overlay* is turned *On*.

Parameters may not be changed while *Overlay* is turned on. However, data files of the same type may be overlaid even if the parameters are different. *Overlay* is turned off if you change the file type.

Note: The print command does not print overlaid curves. It prints the last curve drawn on the screen.

Other Display options—Difference

Sound Lab allows you to difference time and frequency measurements. The *Difference* mode is toggled on or off. When differencing measurements, you first establish a reference curve. The reference curve can be the currently displayed measurement, a recalled file or a new measurement. The reference curve will be subtracted from the next measurement performed and the differenced data will be displayed on the screen. You can also recall a file from the *File...Open Old* menu to difference against the reference.

When using the *Difference* mode, each measurement can be stored on disk under the *File...Save as* sub-menu. When saving the measurement, a sub-menu allows you to save either the (a) *Difference Data* or (b) *Last file or last measurement*. If you choose *Difference Data*, you will save the data as it is currently displayed. If you choose *Last file or last measurement*, you will save the data from the last measurement or recalled file.

When the *Difference* mode is on, *Sound Lab* will not allow parameters of the current measurement type to be changed. If you attempt to make a change, a message box will appear announcing:

Parameters may not be changed while Overlay or Difference modes are on.

If no data exists in memory, you can't turn the *Difference* mode on.



The *Difference* mode lets you remove the frequency response of the loudspeaker when you measure a microphone response:

1. Measure the response of the loudspeaker with a flat-response lab-calibrated microphone.
2. Turn on *Difference* mode.
3. Measure the response of the loudspeaker with the microphone under test.
4. (1) will be subtracted from (3), leaving only the microphone response.

Other Display options—Cursor

Turn the cursor *ON* to read values of the data points on the graph. The *F2* key also turns on the cursor. Move the cursor along the graph by clicking the mouse or pressing the Arrow keys.

The Arrow keys move the cursor as follows:



Move left one data point



Move right one data point



Move right 10% of the display



Move left 10% of the display

Ctrl +



Move 10 data points to the left

Ctrl +



Move 10 data points to the right

Illustration 4-42

Arrow keys that move the cursor

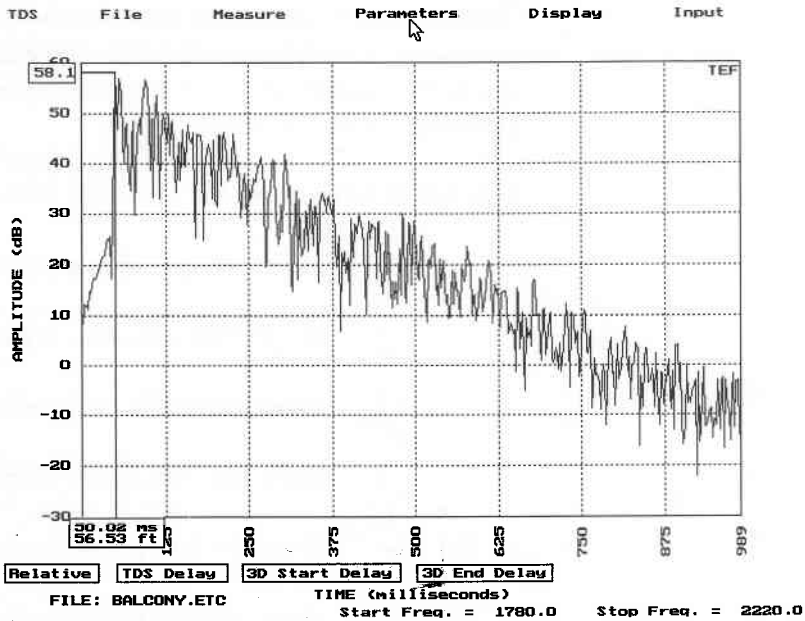


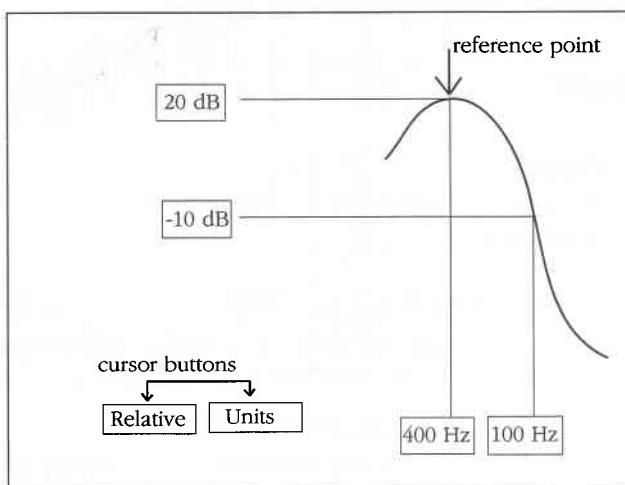
Illustration 4-43
Time response display showing the cursor data windows

Other Display options—Relative cursor

The relative cursor mode is available for both the time and frequency cursors. The relative cursor is used to find curve data values relative to a reference point on the curve. To use the relative cursor, first turn the cursor on and position it on the point you want to designate as the reference point.

Select the relative button at the bottom of the screen (click on it or press R) to make this the reference point. As you move the cursor, you will note that a "phantom cursor" is left behind at the reference point and the cursor values are now referenced to that point. For example, if the cursor is on a data point that is 10 dB below and 100 Hz to the right of the reference point, the value in its data window will read -10 dB and 100 Hz. If you choose the *Units* cursor button, the cursor data window will show the magnitude in your chosen *Reference Units* as set in the *Inpit...Calibration* sub-menu.

Illustration 4-44
Relative cursor



RT60 cursor

The *RT60 cursor* is toggled on or off. The *Shift-F2* key also activates the RT60 cursor.

When the RT60 *cursor* is *On*, *Sound Lab* performs an integration on the time response data and displays it in a second color. Three cursors can then be moved across the data to process RT60 calculations and a %ALCONS calculation. *Sound Lab* displays the RT60, the direct-to-reverberant ratio, the %ALCONS, and the difference in level between the left and right cursors on the integrated curve.

Information boxes

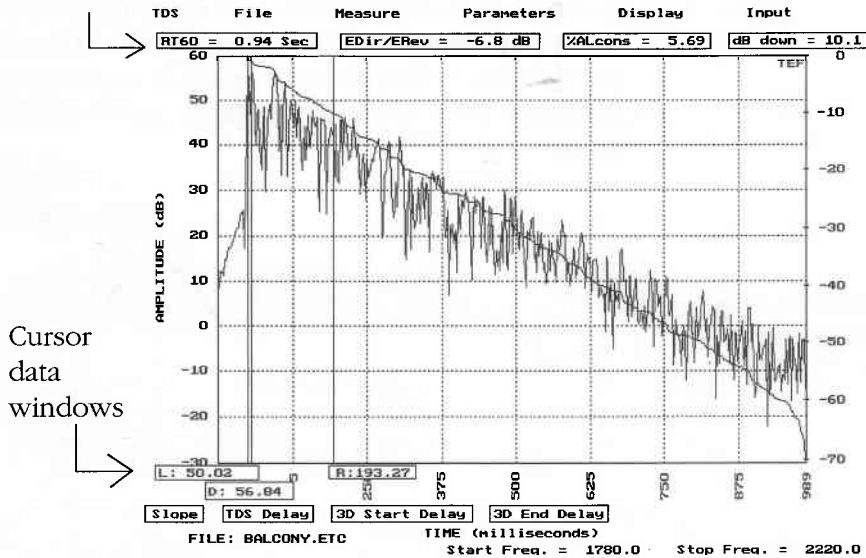


Illustration 4-45

Time response (ETC) display with RT60 cursors active.



As the active cursor moves across the screen, each of these values is updated.

All three cursors can be manipulated with mouse or the Arrow keys

On the ETC graph the cursors select:

- L Left end of the RT60 computation
- D Division between the early and late sound used in the calculation of early-to-late energy ratio for a %ALCONS measurement
- R Right end of the RT60 computation

The text in the data readout of the active cursor is a different color than the other two. To select a different active cursor, press the prefix letter (*L*, *D*, or *R*).

The RT60 cursor is active only for time response data.

Other Display options—Adjust colors

Sixteen colors are available for *Sound Lab* displays and menus. When you select *Adjust Colors*, a list appears showing various display elements along with their current colors. Highlight the item in the left-hand list, then press *Enter* to open another sub-menu with a list of color choices. Make your selection and press *Enter*. *Sound Lab* will update the color sample in the sub-menu.

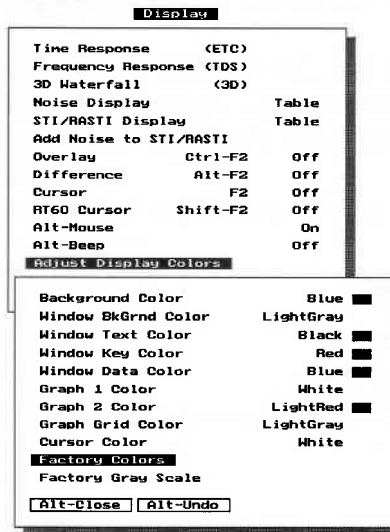


Illustration 4-46
Submenu to adjust display colors.

Sound Lab will not allow color combinations that will cause display elements to disappear. For instance, if *Window background* is black, you may not set *Window text* to black.



Illustration 4-47
Sound Lab display colors



Summary of Display menu options

Top of Scale—The *Top of Scale* value (in dB) determines the value that will be displayed at the top of the vertical scale.

Bottom of Scale—The *Bottom of Scale* value (in dB) determines the value that will be displayed at the bottom of the vertical scale.

Auto Scaling—Toggle *Auto Scaling* on or off. If *Auto Scaling* is *On*, the TEF will automatically scale the data to display the full magnitude range of the measurement. The highest data value is placed in the top 10 dB of the graph, and the scale annotation is adjusted accordingly. The new *Top of Scale* and *Bottom of Scale* will be updated when the data is redisplayed. Manually entering a value for *Top of Scale* or *Bottom of Scale* automatically toggles *Auto Scaling Off*.

We recommend that *Auto Scaling* be left on for most measurement tasks. If *Auto Scaling* is off and all the data is out of the selected range, you will not see any data on the screen.

Auto Scaling can be toggled off if you don't want the graph's *Top of Scale* value to change with data level from test to test. This is a requirement if you are going to use *Overlay* to create a family of curves on the screen resulting from several tests. See additional comments in *Overlay* command description.

Wrap—When *Wrap* is toggled *on*, the phase curve will wrap or start over at the top of the scale as frequency progresses. The top of the phase graph will be 180 degrees and the bottom will be -180 degrees. With *Wrap off*, the phase curve will be continuous, without any sudden 360-degree phase jumps or transitions. If you edit the *Top of Scale* or *Bottom of Scale*, the *Wrap* feature turns off.

Graph—Toggle to choose *dB* or *Linear* data in the Nyquist display.

Display Rotation—Data entry for the degrees of counter clockwise rotation you want in the Nyquist display, starting from zero degrees.

Horizontal Scale—Choose a *Linear* or *Log* display to set the horizontal frequency scale in frequency response displays.

Octave smoothing—Enter a percentage value for amount of octave smoothing to be performed on the data in (a) magnitude and (b) magnitude and phase displays. The value may be entered as a fraction, but it will be displayed in the menu as a percent.

Perspective (3D)—Toggles between *Left* and *Right* views of the display.

Curve Order (3D)—Toggles between *Front to Back* and *Back to Front*, describing the placement on the graph of the first curve measured. The first curve is the one that is measured at the time set by *Start Receive Delay*.

Front to Back (3D)—The first curve measured (curve 001) is placed in the front of the graph.

Back to Front (3D)—The first curve measured (curve 001) is placed in the back of the graph.

NC—Toggle this command to display noise data as a *Table* or *Graph*.

STI—This command toggles the type of display for STI measurements. *Table* displays the STI value for each band plus the overall STI value. *Graph* displays the STI values on a graph.

Overlay—allows multiple time, frequency or NC measurements to be placed on the screen at the same time.

Difference—Time or frequency measurements can be differenced. The Difference mode is toggled *On* or *Off*. After a reference curve is established, it is subtracted from succeeding curves and the differenced data is displayed on the screen.

Cursor—Turn the cursor *On* to read values of the data points on the graph. Move the cursor along the graph by clicking the mouse or pressing the Arrow keys.

RT60 cursor—The RT60 cursor is toggled on or off. When the RT60 cursor is *On*, *Sound Lab* performs an integration on the time response data and displays it in a second color. Three cursors can then be moved across the data to perform RT60 calculations and a %ALCONS calculation.

Alt-Mouse—Turns off the mouse operation.

Alt-Beep—Turns off the beep warning.

Adjust Colors—*Adjust Colors* allows you to choose from sixteen colors for the various elements of the display. Current colors are shown along with a list of color choices.

Input menu- setting the hardware and software to work together

In the *Input* menu, you select the microphone preamp or line level inputs, specify the reference unit values you wish to use, and the computer port that communicates with the TEF.

- To set the input options, pull down the *Input menu* and select *Settings*
- To calibrate the display, pull down *Input* and select *Calibration*
- *Communication* selects communications options.

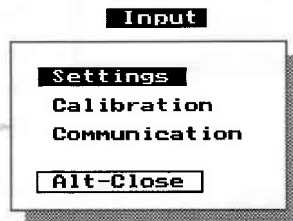
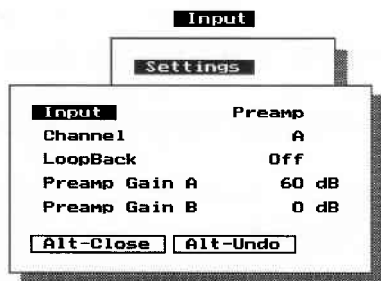


Illustration 4-48 Input sub-menu



Settings sub-menu

Input—Toggles between *Line* and *Preamp*. The TEF has two inputs for each channel: BNC connectors are for line-level inputs and three-pin XLR connectors are for microphone inputs. The microphone input is connected to a microphone preamp to amplify the signal levels.



Channel—Toggles between channel *A* and *B*—the channel on which the TEF receives data. This applies to both line and microphone inputs.

The TEF analyzer can accept input from one of four signal sources (*Channel A-Line* or *Preamp* or *Channel B-Line* or *Preamp*).

Illustration 4-49
Settings sub-menu

Loopback—Toggles between *Loopback On* and *Off*. *Sound Lab TDS* can test the TEF hardware with an internal loopback connection from output to input. To verify that the TEF is working, turn on *Loopback* and perform a frequency response test. The resulting display shows the internal frequency response of the TEF.



Preamp Gain—*Preamp Gain A*: To set the gain of the microphone preamp for Channel A, elect *Preamp Gain A*, and enter the number of dB of gain you wish the microphone preamp to have. A typical value is 40 to 60 dB. Use the same method to set *Preamp Gain B*.

Sound Lab accepts gain changes in 4 dB increments from 0 to 60 dB (0, 4, 8, etc.) When setting the input gain, use as much gain as possible in the preamp (just like in a sound system, put the gain up front). Once you start measuring, adjust the gain up or down until it is as high as possible without getting an overload indication from the software.

If the overload (ovld) LED on the TEF lights during the measurement, either the Preamp Gain or the system output level is too high. A good practice is to set the output level of the amp to obtain a test-signal level of about 70 dB SPL (conversation level). Then, if the overload light comes on, reduce the microphone preamp gain.

Note: It is not a good practice to set the output level with the output level knob on the TEF since some tests require this knob to be in the calibrated position to get calibrated results.

Calibration sub-menu—

Calibrating the display

The amplitude reading you get in a display is always accurate in a “dB relative” sense. If you need to know the absolute amplitude of a measurement relative to 0 dB SPL, the numbers will be incorrect unless you have first calibrated the instrument.

To display data, *Sound Lab* software uses reference units, volts per reference unit, propagation speed, distance units and a 0 dB reference. Without this information, *Sound Lab* would show data only in terms of volts and seconds, and not the more familiar units of sound pressure level and feet.

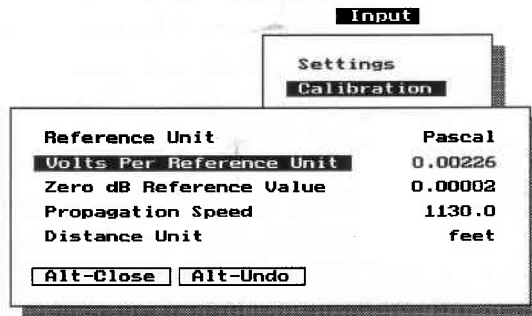


Illustration 4-50 Calibration sub-menu

To enter these units, choose *Input* and select *Calibration*. The first time you use the software, try these values:

Reference unit	pascal
Volts per reference unit00226 (Typically B&K 4007 microphones are .00226 volts/pascal. See your microphone data sheet under the specification <i>Sensitivity</i> .)
Zero dB Reference Value	0.000020 pascal; this corresponds to 0 dB SPL.)
Propagation speed	1130 feet/second (for sound traveling in air at 20°C)
Distance Unit	Set to <i>Feet</i> ; this correlates with the propagation speed. The propagation speed will change as you change the Distance Unit. For example, if distance unit is set at 1130 feet/second and you change to meters, the value will automatically change to 344.4 meters /second.

Table 4-5 Initial calibration settings

Reference Unit—Enter the name (up to 10 characters) for the reference unit you want to use. To set the reference unit, choose *Reference Unit* and enter the new value. The pascal is the standard reference unit for acoustic measuring. If you are measuring voltages, type *Volt*; for impedance measurements enter *Ohm*.

Volts per Reference Unit—Enter the sensitivity value from your microphone or transducer data sheet. This value indicates how much voltage your transducer generates when one reference unit is applied to it.

For measurements with a microphone, this value indicates how much voltage the microphone generates in a sound field of one pascal (or 94 dB SPL). For the B&K 4007S microphone available from Techron, a typical value is 2.26 millivolts per pascal (entered as 0.00226).

If you are measuring electronics, and the reference unit is one volt, type 1.

Zero dB Reference Value — This value indicates the zero dB reference value for your measurements in terms of your chosen reference unit. Choose *Zero dB Reference Value* and type in the value that you want to correspond to the 0 dB line on the measurement graph.

- For acoustics, this value is 20 micropascals, which corresponds to 0 dB SPL. Enter 0.00002 for this value.
- For other measurements, enter the value that yields 0 dB. For example, enter 1 volt for dBV.

Propagation Speed — Highlight *Propagation Speed* and enter the propagation speed for the media in the system you are testing. For example, sound travels 1130 feet per second in air. The number entered represents the distance a wave travels in one second in terms of your chosen distance unit. For sound systems, the distance unit is usually meters or feet. *Sound Lab* software uses the propagation speed setting in several places to convert between time and distance.

Media	Meters/Second	Feet/Second
Air	344	1128
Water (fresh)	1480	4855
Water (salt)	1520	4987
Glass	5200	17060
Gypsum board	6800	22310
Concrete	3400	11155
Wood (soft)	3350	10991
Aluminum	5150	16896
Mild steel	5050	16568
Lead	1220	4002
Plexiglass	1800	5905
Human body	1558	5111

Table 4-6 Speed of sound in various media (at 21°C)



Distance Unit—*Sound Lab* allows four distance units. Usually this unit will be *Feet* or *Meters*, depending on whether you want to use the English or Metric system of measurement. The propagation speed automatically changes to match the units.

When you choose *Distance Unit*, a sub-menu appears allowing you to choose *feet*, *inches*, *meters*, or *centimeters*.

To keep *Distance Unit* setting as displayed, press *Alt-Close*.

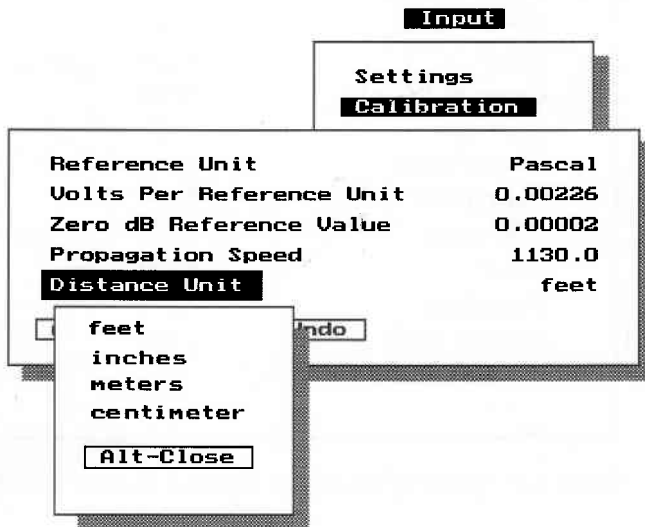


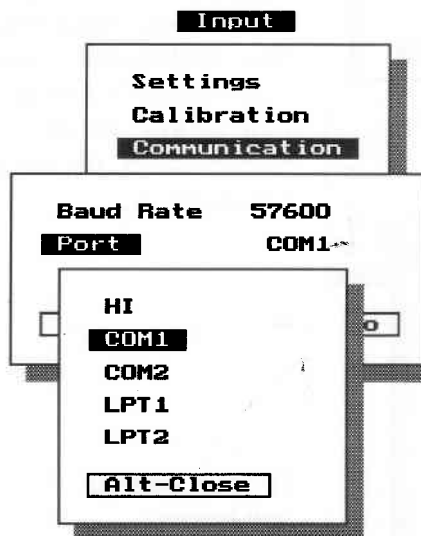
Illustration 4-51 Distance unit sub-menu

Communication sub-menu

Use the *Communication* sub-menu to match hardware and software settings for communication between the TEF and your PC. Sound Lab lets you choose serial (COM1 or COM2), HI (host interface) or parallel communications (LPT1 or LPT2).

To use serial communication:

1. Set Baud Rate to match the serial interface settings made on the analyzer. For more information, see Section 5 in the *TEF 20 Analyzer User's Manual*.
2. Connect the analyzer to a serial port connector designated as either COM1 or COM2 on your PC. See your computer installation manual for the location of COM1 or COM2. One of these ports may be used for the mouse. Make the same selection in the *Communication* sub-menu.

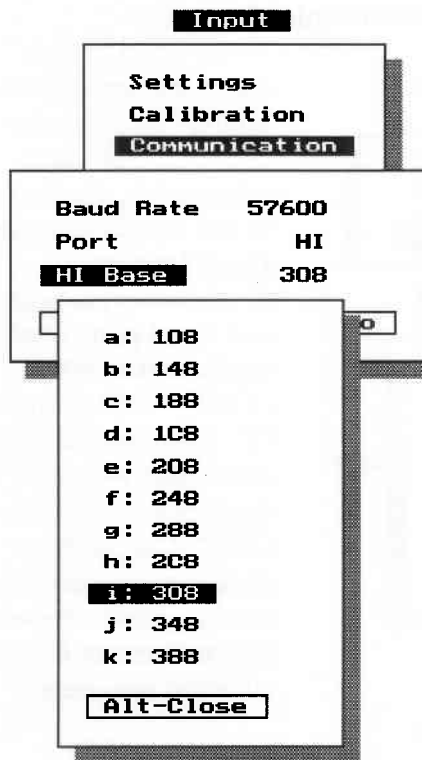


*Illustration 4-52
Setting sub-menu*

To use HI (host interface) communication:

1. Instructions for installing the HI PC card can be found in Section 5, Pages 5-7 to 5-9, in the *TEF 20 Analyzer User's Manual*.
2. Connect the TEF analyzer to your PC with the cable that came with the HI PC card.
3. Select HI in the *Communication* sub-menu.

4. Select the HI Base address with the *HI Base Address* sub-menu. The address set on the HI computer board must match the address you select with the HI Base Address sub-menu. The manufacturer's setting for the base address is 308. To change these settings, see the page reference above.



*Illustration 4-53
HI Base Address
sub-menu*

To use parallel port communication:

To use the parallel port, be sure the parallel cable is connected to your computer and simply choose *LPT1* or *LPT2* to match the port connection on your computer.

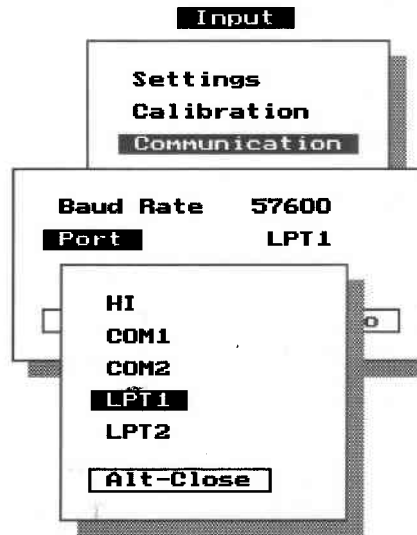


Illustration 4-54

Sub-menu for selecting parallel communications

Section 5: Performing measurements

Introduction

Sound Lab TDS software combines with the TEF analyzer and your computer to form a complete measurement system. The TEF analyzer generates a signal into a device or environment under test, and then retrieves, analyzes and visually displays the many Time, Energy, and Frequency relationships of the associated data.

Sound Lab performs traditional measurements such as the time and frequency response, Noise tests, STI/RASTI intelligibility tests, and 3D tests. Many options for display of the data are easily accessed from the Display menu.

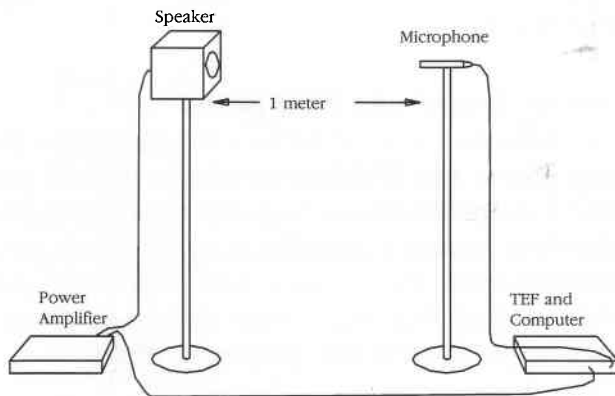
To make Sound Lab TDS tests

This section of the manual reviews the basic steps for making *Sound Lab TDS* measurements. Before you attempt these measurements, be sure you are thoroughly familiar with *Section 3, Navigation*, which details how to navigate around the program, work with menus and enter parameters. For more details about the software and setting parameters, see *Section 4, Menus*.

To make *Sound Lab TDS* measurements, you may need an MS-DOS-based computer, a TEF 20 analyzer, *Sound Lab TDS* software, an amplifier, a speaker, a tripod and a high quality microphone. An alternative to using a microphone is a sound level meter with a line-level output. See *TEF System 20 Analyzer User's Manual* for hardware details.

Performing a time response test on a loudspeaker

1. Connect the system as described in the *TEF 20 Analyzer User's Manual*. Put the loudspeaker and microphone on stands, several feet from any reflective surface. Put the microphone 1 meter



from the speaker, on axis. On the TEF analyzer, set the output level knob about 1/4 of the way up. Set your power amplifier level control about 1.2 of the way up. These low settings prevent a loud burst of noise when you run a test. A test signal that is too loud can damage a speaker or cause distortion.

Illustration 5-1
Equipment arrangement for performing
Time and Frequency test of a loudspeaker.



2. Start the *Sound Lab* program.
3. Set the input parameters. Go to the *Input* menu and select *Settings*. Select the settings and gain to match your equipment arrangement.
4. From the *Input* menu, select *Calibration* and enter the proper values and calibration constant for your transducer.
5. From the *Input* menu, select *Communication* and enter the proper values.
 - a. If you do not have a TEF-HI card installed, enter 57600 for the *Baud Rate* of the *com port*. If you do have a TEF-HI card installed, the baud rate doesn't matter.
 - b. If you have a TEF-HI card installed, select *HI port*. Otherwise, select *COM1* or *COM2*, depending upon the port to which you connected the TEF.
 - c. Use the *HI Base* default—308.
6. Setting the Display. From the *Display Menu*, choose *Time Response*, then from the *Display Type* submenu, select *Magnitude*. Turn *Auto Scaling On*.
7. Setting the measurement parameters. Go to the *Parameters* menu, and select *Time Response* test. Enter the parameters under which you want the test conducted. For more information on setting parameters, see *Section 4, Menus, Guidelines for setting parameters*

8. At this point you may run the test by selecting *Do Time Test* under the *Measure* menu, or by pressing function key *F5*. You will see the display on the screen and be able to examine it.

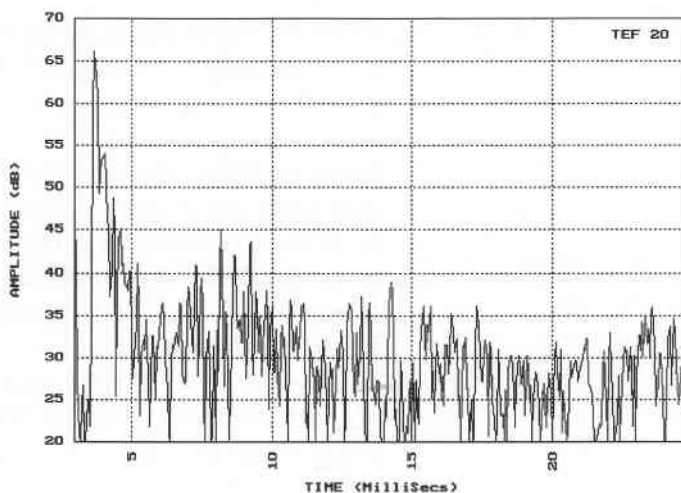


Illustration 5-2
ETC Display of a Time Response Test

9. Refine the time scale if the first reading is too coarse for clarity. See *Appendix A, How TEF works* for more information on resolutions.
10. From the *Display* menu, select *Time Response* to experiment with other ways to display the data.

11. Printing the display. If you wish, you can print the graph. From the *File* menu, choose *Printer Settings*. Try using the default settings. With your printer turned on, select *Print* to print the display.
12. Saving the settings. You can save any settings you used in testing and load them later by selecting *Configuration* and entering a name in the dialog box which appears.
13. Saving the data. To save the test data, select *Save As* from the *File* menu. Select *Clear All* to set up the dialog box for your information. Enter *File Name* and any other information you desire. Select *Binary* under *Format*, select *Save* and close the dialog box to continue with your next measurement.

Performing a frequency response test

1. See equipment arrangement for Time Response test.
2. Start the *Sound Lab* program.
3. Set the input parameters. Go to the *Input* menu and select *Settings*. Select the settings and gain for your equipment arrangement.
4. From the *Input* menu, select *Calibration* and enter the proper values and calibration constant for your transducer.
5. From the *Input* menu, select *Communication* and enter the proper values.
6. Perform a time response test (ETC) to find the direct sound.
7. Turn on the *Cursor* in the *Display* menu by selecting *Cursor On* or pressing function key *F2*. Position it over the exact time of the first energy arrival at the microphone. Press *F4* or select the *TDS Delay* button at the bottom of the screen to enter the receive delay into your measurement parameters for the Frequency Response test.
8. Setting the measurement parameters. From the *Parameters* menu, select *Frequency Response*. Enter the parameters under which you want the test conducted.

9. At this point you may run the test by selecting *Do Frequency Test* under the *Measure* menu.
10. Refine the receive delay to flatten the phase response by pressing *F4* and entering the incremental changes you desire. Press *Enter* and retest.
11. From the *Display* menu, select *Frequency Response* to view the data in other displays. You can choose *Phase*, *Magnitude*, *Magnitude and Phase*, *Nyquist* and *Heyser Spiral* displays.

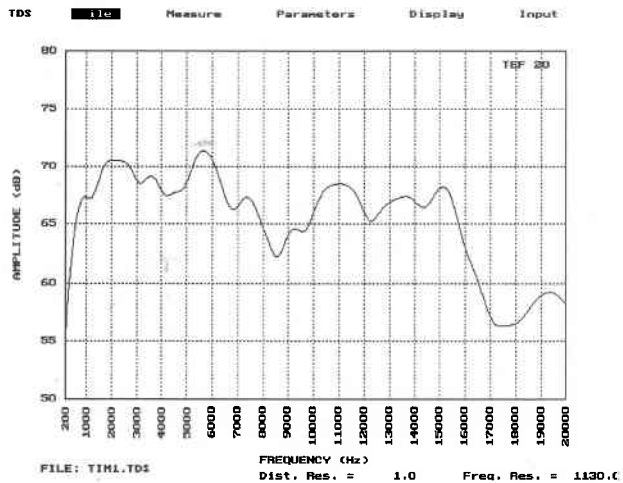


Illustration 5-3
Magnitude Display of the Frequency Response test.

Noise criteria measurements

NC (Noise Criteria) Curves are a standard method of characterizing noise loudness. The NC rating has been used extensively in noise and vibration measuring. Noise Criteria measurements (NC) can provide important information for designing a sound system or evaluating a noise problem. NC measurements are made with ANSI Type 2 filters.

Equipment needed

To make NC measurements you will need an MS-DOS computer, a TEF 20 analyzer, *Sound Lab TDS* software, a tripod and a high quality microphone. An alternative to using a microphone is a sound level meter with a line-level output.

NC measurements with a microphone

The microphone inputs of the TEF 20 are wired to accept industry standard 3-pin XLR-type connectors. The microphone connector provides phantom power for condenser microphones.

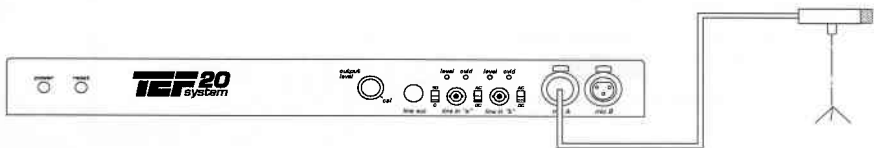


Illustration 5-4 Equipment arrangement for microphones

If you are using the built-in TEF preamps, adjust the gain in the preamp until you get an overload, then lower it one step to get the proper level.

Note: Be aware that all microphones have inherent self noise, and if you are measuring an extremely quiet room, the self noise of the microphone could adversely affect your measurement. Check your microphone specifications for this information. It may help to have your measurement setup in a different room from where the NC is being measured.

NC measurements with sound level meters

If you are using a sound level meter and the line level input, the output of the sound level meter must be in the range of 1 millivolt to 1 volt to perform valid NC measurements. To connect a sound level meter, connect the output of the meter to the TEF 20 line input and set the AC/DC switch to AC.

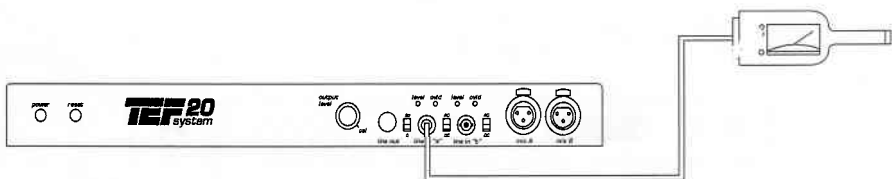


Illustration 5-5 Equipment arrangement for sound level meters

Setting the *AC/DC* switch to *AC* will prevent small dc voltages (generated by the sound level meter) from causing premature and otherwise unexplained input-overload errors.

Additionally, be sure to understand how the meter range settings affect the output level. Consult the technical manual supplied with the sound level meter before attempting an accurate calibration

Making NC measurements

This section details the basic steps for making an NC measurement. For more details about the software, see the section titled *NC menus*.

To make an NC measurement, follow these steps:

1. Choose *Settings* from the *Input* menu. Choose the input setting and gain for your equipment arrangement. To insure accurate readings, adjust the preamp gain until you get an overload, then lower it one step to arrive at the proper level.
2. Choose *Calibration* from the *Input* menu. Enter the calibration constant for your transducer.
3. In the *Parameters* menu, choose *Noise Test (NC)* and enter a value between 1-9999 seconds for the *Integration Time*.
4. Choose *Do Noise Test (NC)* from the *Measure* menu, or press *F7*.

NC will start the test and measure the noise in eight octave bands. When the test is complete, the data will be displayed on the screen.

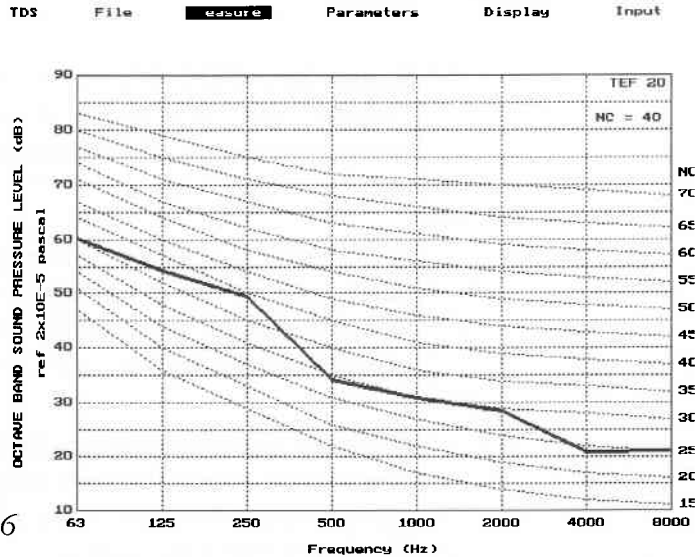


Illustration 5-6
NC curve

The single-number NC rating assigned by NC is the highest curve that is not exceeded by data in any octave-band. This implies that noise in a single octave determines the NC rating since the effects of the different bands are not additive.

The lower boundary of the family of NC curves (NC15) represents the point at which noise is perceived by most people to be objectionable. The highest curve (NC70) represents noise in which people can communicate only by shouting in each other's ears.

The curves are spaced 5 dB apart through the mid-frequencies because the average listener will not react to a change in the noise that is less than 5 dB (even though smaller changes can be detected.) The shape of the curves accounts for low frequency noise being less annoying (or noticeable) than high frequency noise.

Introduction to STI

The speech transmission index (STI) is an evaluation of how well the amplitude modulation patterns of speech are preserved when passed from their source to a listener.

STI and RASTI measurements

Sound Lab TDS makes two types of speech intelligibility measurements based on the Speech Transmission Index (STI).

- STI measures in the octaves from 125 Hz to 8 kHz.
- RASTI, a shortened version of the STI procedure, measures only at the 500 Hz and 2 kHz octaves.

Sound Lab STI collects data from a noise measurement and several time response tests which are processed to yield an overall speech intelligibility prediction. The resulting data is displayed on the screen in a graph or table. The software default measures with noise included, although it can be turned off. Practical STI/RASTI results, however, have noise data added into the calculation, and *Sound Lab TDS* allows you to add noise later.

Sound Lab TDS allows you to make STI measurements by using a calibrated talker/speaker placed in the location and position where a human speaker would be. This talker/speaker must be flat (under free-field conditions) within plus or minus 10 dB between 8 kHz and 11.2 kHz. A test microphone is placed in the position of the listener.

For valid speech intelligibility results, the talker/speaker should be calibrated to a level that simulates the voice level of a person speaking and equalized to eliminate any deficiencies of the speaker. This operation precedes the STI test.

The *Calibration* procedure is set up in the *Parameters...STI* sub-menu and the *Speaker EQ test* option automatically follows calibration. The calibration and equalization tests automatically precede the STI test when you choose *Do STI Test* under the *Measure* menu. A series of dialog boxes will lead you through the procedure, and then the software will proceed directly into measuring the STI.

See *Calibration* procedure to follow.

Performing the measurement with calibrated talker/speaker

This section details the basic steps for making an STI measurement with noise included and assumes you are going to perform the accompanying calibration and equalization tests of the talker/speaker. To make an STI measurement, follow these steps:

Equipment arrangement for calibration

1. Place test microphone and talker loudspeaker one meter apart, on axis. An alternative distance placement is 1/2 meter apart.
2. Connect the input of the test amplifier to the test output of the TEF. Connect the output of the test amplifier to the talker loudspeaker. Connect the test microphone to either channel A or B of the TEF.

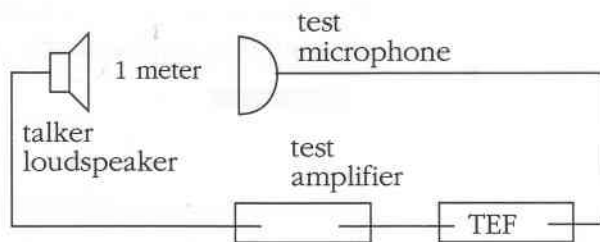


Illustration 5-7

Typical equipment arrangement to perform STI calibration procedure.

C

- 

Measure

Testing

Starting Source Calibration procedure.
Adjust until level is 61 at 1 meter.
(67 at 0.5 meter)
Current level is 75.0
Press Esc to continue...

C

6. As you monitor the level on screen, manually adjust the gain of the test amplifier until the value in the dialog box reads 61 (67 for 1/2 meter test.)
7. After the correct level is attained, press *Escape* to continue to the Equalization procedure. A message box will appear with the message:

Testing Speaker EQ

The TEF is running a standard frequency response test from 88 Hz to 11.3 kHz (one-half octave below 125 Hz and one-half octave above 8 kHz), and then stores the measured equalization value.

Note: For all subsequent intelligibility tests, keep the gain control of the test amplifier at the setting that you determined in this calibration procedure. If the gain control of the amplifier is changed, perform the calibration procedures again.

Following the EQ test, the test is paused and a dialog box informs you:

- *Source Calibration* procedure is completed
- *Source Calibration* is being turned off
- You may arrange equipment for the STI/RASTI test.

You continue the procedure by pressing any key. See *Illustration 5-10 below..*

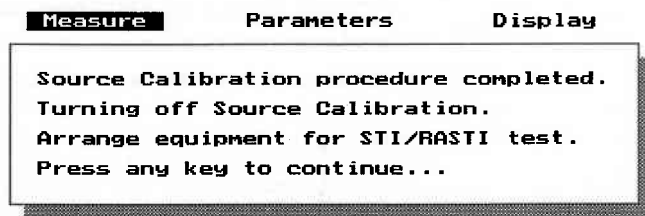


Illustration 5-9
Dialog box preceeding STI sweeps.

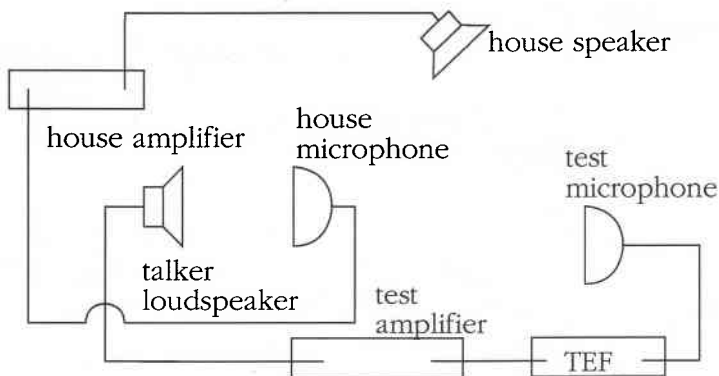
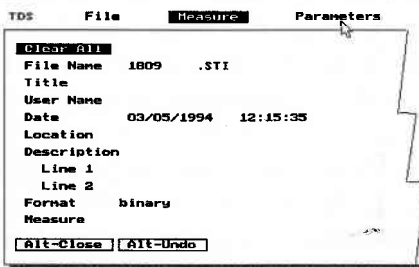


Illustration 5-10 Equipment arrangement for STI/RASTI test

Arrange the equipment for STI

1. Place the source loudspeaker where the orator will speak and adjust the height of the loudspeaker to be about head high.
2. Place the test microphone in the position of a human listener.
3. Turn on the house system and set the levels for normal operation
4. Press any key to resume the STI/RASTI tests.



*Illustration 5-11
Save As dialog box allowing
you to name the test and start
the STI sweeps.*

Pressing any key causes a *Save As* dialog box to pop up allowing you to name the STI measurement and create testing notes before continuing. You must choose *Measure* from the dialog box to start the test.

Sound Lab will continue the STI/RASTI test and first measure any noise in eight octave bands. When the noise tests are complete, STI then performs seven ETC tests (two for RASTI). When all of the ETCs are complete, the software then calculates the over-

all STI and displays the results in the active window in a table or graph, depending on your choice made under *STI/RASTI* in the *Display* sub-menu.

Since calibration and equalization are complete, to measure at other listener positions, you can simply initiate the *STI/RASTI* sweeps from under the *Measure* menu in your usual manner. The level calibration and equalization procedure should be performed every time a different loudspeaker or microphone is used and any time the gain controls are altered in the measurement setup.

Driving the system directly

If you do not use the talker/speaker arrangement, but prefer to go through the house system directly, the levels again should be set the same as for a person speaking. In this arrangement, of course, you do not perform the calibration procedures. Make certain that the reinforcement system level is predominant either by listening or by making a Time Response test and checking the signal strength of the reinforced sound and the time of arrival.

Adding noise to STI measurements

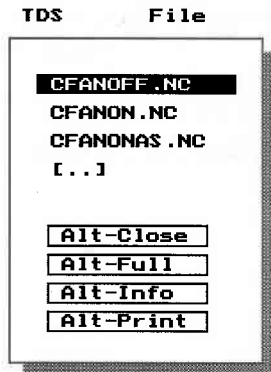
Noise can be added to STI measurements from an NC (noise criteria) measurement or from a table. The *Add Noise to STI/RASTI* option in the *Display* menu provides a way to post-process noise with noiseless STI or RASTI data.

When you have STI data displayed on the screen, choosing *Add Noise to STI/RASTI* opens a sub-menu that allows you to choose noise values from an NC file or Table.

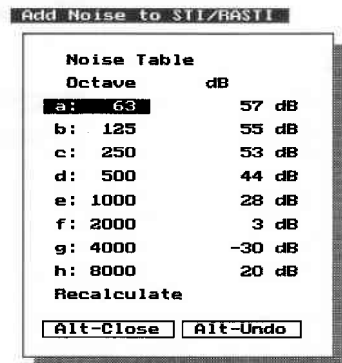
If you choose *NC File*, a sub-menu opens showing available files from which to extract noise data. If you choose *Table*, a sub-menu will appear allowing you to enter noise values on octave band centers from the keyboard.

The software will perform an FFT on the data with the noise included and redisplay the results.

Illustration 5-12
Noise can be added to STI/RASTI measurements from a file (A) or a table (B) under the *Display* menu.



A.



B.

Evaluating the STI graph and summary table

You may choose between two displays – a graph or a table display.

An STI graph plots the individual STI scores in octave bands.

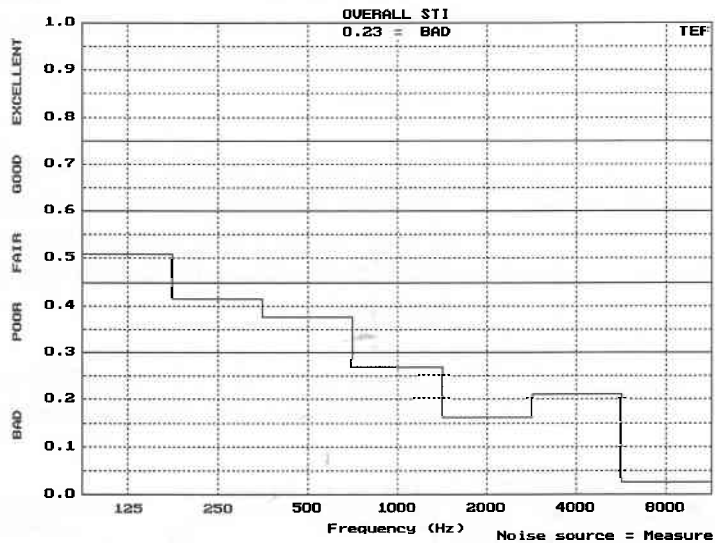


Illustration 5-13
STI data displayed as a graph.

STI summary table display

The STI table display shows numeric values of the individual and overall STI data. Shown are the STI scores, the equivalent early reverberation time and the equivalent signal-to-noise ratio (S/N Ratio).

TEF SPEECH INTELLIGIBILITY TEST				TEF 20
FREQUENCY	STI	EARLY RT60	S/N RATIO	
125 Hz	0.51	1.51	0.3 dB	
250 Hz	0.42	2.44	-2.5 dB	
500 Hz	0.38	2.99	-3.7 dB	
1000 Hz	0.27	5.21	-6.9 dB	
2000 Hz	0.16	10.77	-10.2 dB	
4000 Hz	0.21	7.03	-8.6 dB	
8000 Hz	0.02	47.97	-14.3 dB	
-- OVERALL --				
STI = 0.23				
Equivalent S/N Ratio = -6.6 dB				
Equivalent Early RT60 = 11.13 s				
-- SUBJECTIVE EVALUATION --				
BAD				

File: BALCONY.STI

Noise source = Measure

Illustration 5-14
Summary STI table display

The subjective evaluation—The STI subjective score correlates with spoken word articulation tests. These tests are conducted with a speaker reciting a phonetically balanced word list for a panel of listeners who record what they hear.

Subjective evaluation	STI score		
Bad	0	to	0.3
Poor	0.3	to	0.45
Fair	0.45	to	0.6
Good	0.6	to	0.75
Excellent	0.75	to	1.0

Signal-to-noise ratio—This column indicates the signal-to-noise ratio that would yield the STI score if no reverberation were present. This assumes that only noise, without the effects of reverberation, is present in the transmission chain to degrade intelligibility.

Early RT60—The early RT60 column indicates the value of the early decay time that would yield the STI score if no noise were present. This assumes that only reverberation effects are present in the transmission chain to degrade intelligibility. The reverberation process assumed in this calculation is a perfectly linear decay (on a dB basis) but with no noise.

*Table 5-1
STI/RASTI to
ALCONS
conversion table*

Converting RASTI measurements to % AL _{CONS}		
	RASTI	%AL _{CONS}
	0.20-0.22	57.7-51.8
Bad	0.24-0.34	46.5-27.0
Poor	0.36-0.50	24.2-11.4
Fair	0.52-0.64	10.2-5.3
Good	0.66-0.86	4.8-1.6
Excellent	0.88-1.0	1.4-0.0

About the STI measurement process

The modulation transfer function (MTF) is a measure of the preservation of speech modulation patterns necessary to maintain high speech intelligibility when a signal passes through a system. Classical methods directly measure the MTF with sinewave amplitude modulation of a band-limited, random noise, carrier signal. These methods make a series of modulation transfer measurements with carrier frequencies on octave centers from 125 Hz to 8 kHz. The carriers are modulated with speech modulation frequencies of 0.5 to 16 Hz, at one-third octave intervals.

These measurements result in a data set containing 98 modulation reduction indexes which are, in turn, converted into a single index called the Speech Transmission Index (STI). The STI value is a single number that indicates the effect of a transmission system on speech intelligibility.

An alternate method for computing the MTF measures the impulse response of a system without having to directly measure the modulation transfer characteristics at each individual modulation frequency. This method derives the MTF by calculating the frequency spectrum of the squared impulse response of the system.

TDS measurements are a slightly modified version of the impulse response method. Because the TDS process for the ETC measurement yields the complete analytic signal response (both real and imaginary parts) of the system under test, this complete signal information is used to calculate the MTF.



Overall STI and RASTI values are derived by making MTF measurements at all the octave bands between 125 Hz and 8 kHz for STI and at only 500 Hz and 2 kHz for RASTI.

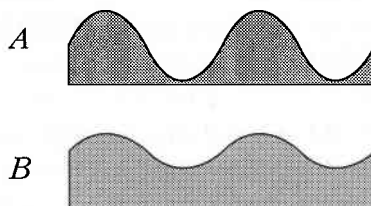


Illustration 5-15

Diagram A shows a sinewave modulating random frequencies which are characteristic of the human voice. This modulation pattern is an example of perfect modulation which equals an MTF of 1.0.

The modulation pattern in Diagram B illustrates the effects of noise and reverberation in the room. The modulation in the received sinewave is reduced by comparison, because of the effects of noise and echoes.



Introduction to 3D measurements

SoundLab 3D measurements made by the TEF analyzer are composed of 2 to 100 individual TDS frequency response measurements. Each sweep is offset in time by a constant amount. When all of the individual sweeps are displayed on the screen, they form a three dimensional display that plots the frequency and time behavior of a system. A 3D measurement shows how the frequency response of a system changes as the system response decays.

If you have not made frequency response measurements, we recommend that you become familiar with the material in this manual on making frequency response measurements.

Use the 3D display to simultaneously examine the time and frequency characteristics of rooms, loudspeakers or electronic devices such as equalizers and filters. For example, you can see frequencies that ring; they have a longer decay time than other frequencies.



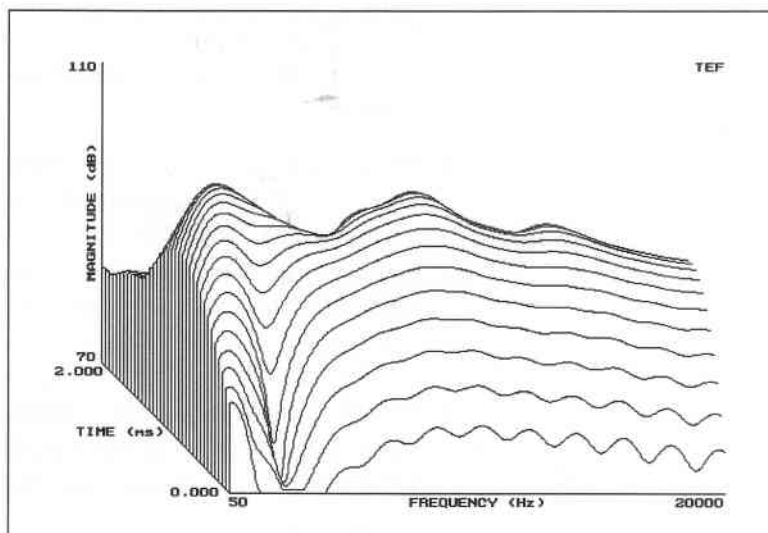
Measuring loudspeakers with 3D

Arrange the measuring microphone and loudspeaker in an area that is free of nearby objects. The distance to the nearest object will determine the lowest frequency you can measure (see *Space window considerations* in Appendix B, *How TEF works*). Position the microphone on-axis with the loudspeaker. The following suggestions assume a distance of five feet between microphone and loudspeaker

1. *Make an ETC.* Set the ETC parameters for a short time span (about 20 milliseconds) to determine arrival of the direct sound and the first reflection and when the sound from the speaker decayed into the noise of the room.
2. *Set the Start Delay.* Turn on the RT60 cursor (Shift F2) and press L to make the left cursor active. To set the start delay, position the left cursor just before the direct sound arrival and enter the value into the 3D parameters by pressing S.
3. *Set the End Receive Delay.* Make the right cursor active and set the *End Delay* by placing it at the point where the direct sound decays into the noise or just before the arrival of the first reflection—whichever occurs first. Press E to enter the value into the 3D parameters. The time between the *Start* and *End Receive Delay* is typically 2 or 3 milliseconds when measuring a tweeter or high frequency horn.

4. *Set the Time Resolution.* In the *Parameters* menu, under *3D Test*, set *Time Resolution* to a value five times longer than the *Receive Delay Step* (set by the start and end receive delay). The ratio of time resolution to *Receive Delay Step* is called the overlap ratio.
5. *Perform the measurement .* Start the 3D measurement by pressing *F9* or *Quick key 3*. Examine the display, and repeat the measurement with different parameters, if desired.

*Illustration 5-16
A Sound Lab 3D
display of a small
speaker system.*



Measuring rooms with 3D

When measuring a room with *3D test*, excite the room in a manner typical of its use. For example, suppose you want to examine the low frequency behavior of a concert hall. Set the loudspeaker in the location where bass sound originates, and place the microphone in the listener's position. Set the frequency span of the measurement to cover the range of the bass frequencies of interest.

1. *Make an ETC.* Set the parameters for a long time span (about 5 to 10 times longer than the time it takes for sound to travel from front to back in the room). Use the cursor to examine the ETC and determine when the direct sound arrived and where the sound decayed into the noise of the room.
2. *Set the Start Receive Delay.* Turn on the RT60 cursor (Shift F2) and press *L* to make the left cursor active. Position the left cursor to the point where the direct sound begins to rise and enter the value into the 3D parameters by pressing *S*.
3. *Set the End Receive Delay.* Press *R* to make the right cursor active. Move it to the right to where the sound decays into the noise.
4. *Set Time Resolution.* In the *Parameters* menu, under *3D Test*, enter a value five times longer than the *Receive Delay Step* in small rooms and to the same value as the *Receive Delay Step* in

large rooms. The ratio of time resolution to the space between sweeps is called the overlap ratio. As explained later, these settings will give a smooth 3D display with good frequency and time resolution.

5. Perform the measurement and examine the display. If desired, repeat the measurement with different parameters.

Overlap ratio in 3D displays

For each offset in a 3D measurement, a single TDS sweep is performed having a time window as determined by the sweep rate and IF bandwidth.

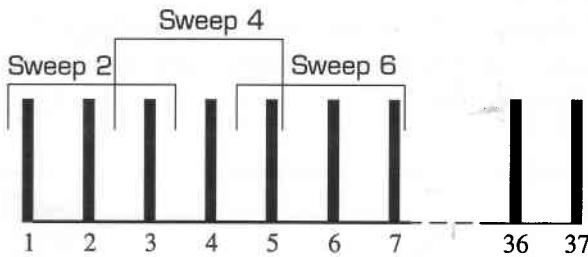


Illustration 5-17 In this illustration, the wide vertical lines represent the time offsets of the individual TDS sweeps. The overlap ratio in 3D measurements refers to time resolution settings that are longer than the time spacing between curves. In this example, the time resolution of each sweep is three times longer than the spacing.

In most cases, the time resolution of each curve will be longer than the time spacing between the curves. The ratio of the time resolution to the receive delay step is called the overlap ratio. The overlap of the TDS sweeps is used to obtain a good compromise between time resolution and frequency resolution. It will also produce a 3D with a smoother appearance.

A good starting overlap ratio for loudspeaker measurements is 5 to 1. That

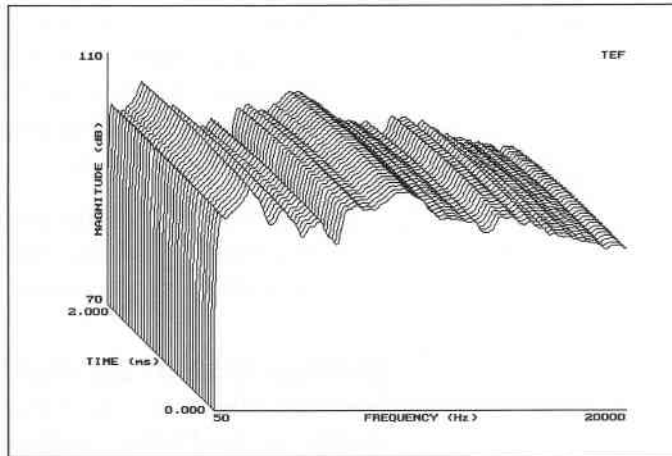


Illustration 5-18

A 3D display with low time resolution and high frequency resolution. It shows good frequency detail, but very little time information.

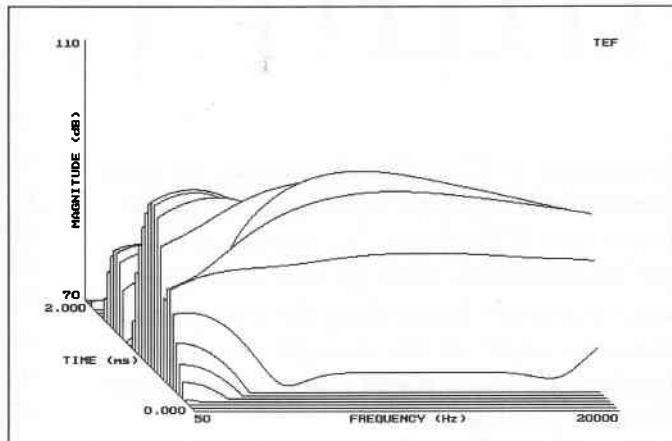


Illustration 5-19

A 3D display with excess time resolution. There is excellent time resolution, but very little frequency detail.

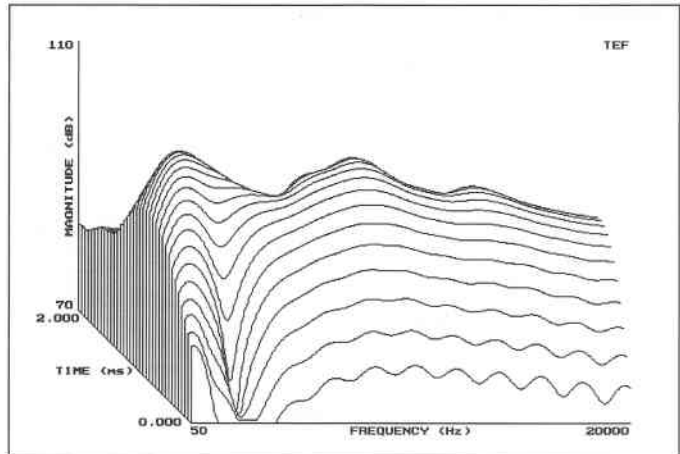


Illustration 5-20

A 3D display that shows a good compromise between the two previous examples. In this measurement we can see detail in both time and frequency data.

Section 6: Practice measurements

Two basic tests

This section of the manual details the sequence necessary for making two fundamental TEF measurements in a typical setup, and gives suggested starting parameters. The object of this section is to help you to design a simple experiment to produce these measurements and get typical displays on the screen. For information on how to enter data or use the menus, see the *Menu* section.

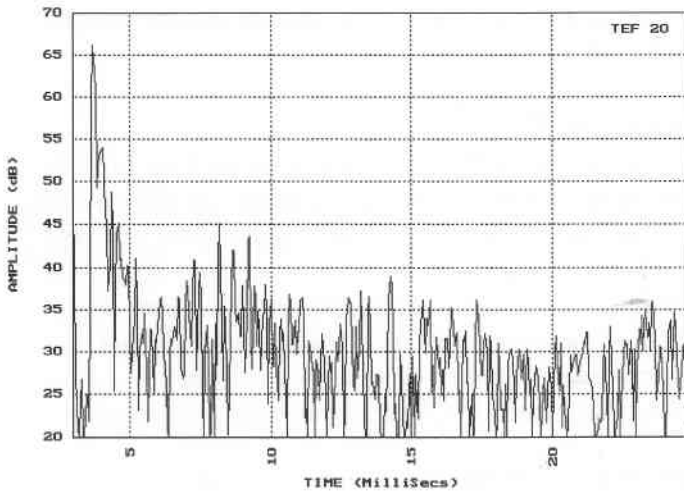
About the ETC

Two fundamental TEF measurements in TDS are the Time Response (ETC) and the Frequency Response. The Time Response test is a fundamental data gatherer for many other TEF measurements. Time Response information is used in setting parameters for Frequency Response measurements and in calculating intelligibility information. For these reasons, it is customary to make a Time Response (ETC) measurement the first task.

Time response measurements are used to:

- set up the frequency response test
- measure the delay of a signal applied to a system under test
- find the direct sound arrival in a room
- find reflections
- find reflections that fit a pattern, such as flutter echoes
- observe the decay rate of sound in a room
- check the coverage in a room
- calculate the RT60 and %ALCONS of a room

The Time Response test displays an Energy Time Curve that shows how energy from a system or device is released after it is excited with a sudden application of input energy confined to a given frequency band over a certain time span. The results are displayed on an ETC graph with time shown on the horizontal axis, and energy on the vertical axis.



An ETC shows the amplitude of the sound energy that arrives at any instant in the time span. The Time Response quickly reveals not only the amplitude and time of arrival, but also the density of the field, its approach to exponential growth and decay, and the initial signal delay.

Illustration 6-1 is a typical ETC. The tall "spike" near the left edge is the direct sound from a test loudspeaker. The height of the spike is not the highest point on the loudspeaker's frequency response curve. It represents the total energy arriving at that particular time—in this case, from the loudspeaker. Further to the right (later in time) reflections may be seen arriving. They have a lower total energy than the direct sound due to absorption and inverse square law losses.

To make Time Response (ETC) tests, the TEF analyzer sends a frequency sweep signal through the system under test (in this case the system is a speaker, microphone and air). The electronic sweep tone fed to the loudspeaker will, in effect, represent all possible frequencies in the chosen sweep range. The TEF analyzer then listens for the sweep via a microphone. The frequency range of the sweep, the sweep rate, and the delay time, plus other factors set by the operator, all determine the characteristics of the ETC test display.

Reading an ETC display

The first tall peak from the left represents the direct sound from the loudspeaker. This is the sound that has traveled directly from the loudspeaker to the microphone in a straight line without having been reflected by some object. The horizontal position of the direct sound on the display indicates its time of arrival (the horizontal scale is time in milliseconds). Its height indicates its level (the vertical scale is amplitude in dB). Immediately following the direct sound in Illustration 6-1, and continuing out to about 5 milliseconds is the decay of the loudspeaker. The various peaks extending from about 5 milliseconds out to the right side of the display, represent reflections in the room.

The *time span* on an ETC display (full scale from left to right) is dependent on the *frequency span* of the sweep and the *number of samples*. The computer changes the *start* and *stop frequencies* to provide the *frequency span* that, combined with current *number of samples* selected, provides the requested *Time Span*. Remember the inverse relationship between the units of time and frequency; a value that is wide in one domain will be narrow in the other domain. A very narrow pulse in the

time domain results in a very wide spectrum in the frequency domain. A wide sweep in the frequency domain results in a short full-scale time in an ETC. For more information, see *Appendix A: How the TEF Works*.

You set the *time span* for ETCs in the *Parameters—Time Response* menu. When you enter a *time span*, the computer automatically changes the *start* and *stop frequencies* to provide the requested *time span*. When long time spans are needed—as when measuring the acoustics of a reverberant room—you will sweep narrow frequency bands. When short time spans are needed—as when examining the fine detail in a loudspeaker's time response—you will sweep wide frequency bands. Keep in mind the inverse relationship between Time and Frequency.

Doing a Time Response Test (ETC)

Equipment arrangement

The physical arrangement for these measurements consists of a room approximately 15' x 10' with a hard-surface folding table, a 4-inch single driver speaker and a B & K 4007 measurement microphone. (You should get similar results with any convenient, single-driver speaker.) The microphone and speaker are mounted on stands approximately 1 1/2 feet above the table, 5 1/2 feet apart, on axis. Connect the output of the TEF to the input of a power amplifier. (Keep in mind that the TEF 20 will output a one-volt signal when the knob is set to the *cal* position.) Connect the output of the amplifier to the loudspeaker. Plug the measuring microphone into the *mic A* connector of the TEF.

Start the *Sound Lab TDS* program. *Sound Lab TDS* will remember the settings that you used when you last exited the program, but if you have not yet used the program, the default settings will appear. To become familiar with measurement parameters, we are going to set each parameter for an ETC measurement.

Setting *Sound Lab TDS* software to work with a microphone requires setting the input hardware and calibrating the display. The *Input* parameters menu includes machine settings that govern the operation of the TEF measurements. Settings for the TEF input vary with the type of microphone and the type of system the TEF is analyzing.

Setting the input parameters

➤ Go to the *Input* menu and select *Settings*.
Set your TEF to the settings shown:

➤ Return to the *Input* menu.

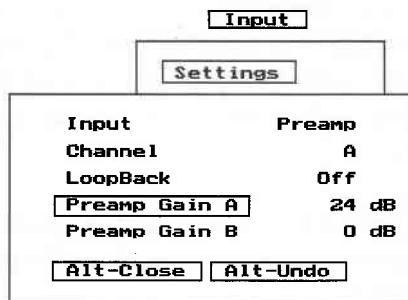


Illustration 6-2
Input parameters sub-menu

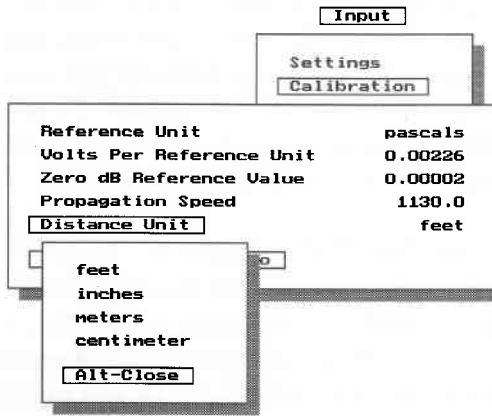


Illustration 6-3
Calibration sub-menu

Calibrating the display

The amplitude reading you get in a display is always accurate in a "dB relative" sense. If you need to know the absolute amplitude of a measurement relative to 0 dB SPL, the numbers will be incorrect unless you have first calibrated the instrument. *Sound Lab TDS* software uses *Reference Units*, *Volts Per Reference Unit*, *Zero dB Reference Value*, *Propagation Speed*, and

Distance Units to show data in terms of sound pressure level and distance. Without this information, *Sound Lab TDS* would show data only in terms of volts and seconds.

➤ To calibrate the display, select *Input...Calibration* and enter the values shown in Illustration 6-3. The *Reference Unit* will reflect the units entered in the edit field, i.e., *pascals*.

The *Volts Per Reference Unit* value is different for each microphone. Please see your microphone data sheet for the correct value. The *Zero dB Reference Value* of .00002 will calibrate the display for sound pressure level.

The default for the *Propagation Speed* is 1130. Changing the *Distance Unit* will update the propagation speed accordingly.

➤ Close the *Calibration and Input* menu; return to the *Main* menu.

Setting the parameters for the Time Response test

► Go to *Parameters* menu and select *Time Response* (ETC). Enter the following settings:

Parameters

Time Response (ETC)

Start Frequency	200.0 Hz
Stop Frequency	10000.0 Hz
Center Frequency	5100.0 Hz
Frequency Span	9800.0 Hz
Sweep Time	1.0 s
Receive Delay	0.0000 ms
Time Span	22.2041 ms
Number of Samples	512
Window	Hanning

Illustration 6-4
Setting parameters
for Time Response
test.

The *center frequency* will be automatically set by the computer when you select the *start* and *stop frequencies*.

The *time span* will be automatically set by the *number of samples*.

Note: The *time span* of 22.2 milliseconds means that the display following the Time Response test will read from 0 milliseconds on the left edge to 22.2 milliseconds on the right.

► Close the *Parameters menu*.

Setting up the screen display

- Open the *Display* menu, and select *Time Response*. Select *Magnitude* under *Display Type*, and turn *Auto Scaling On* if it is not already on. The *Auto Scaling* feature insures that the screen fills with the data you have selected.
- Close the *Display* menu.

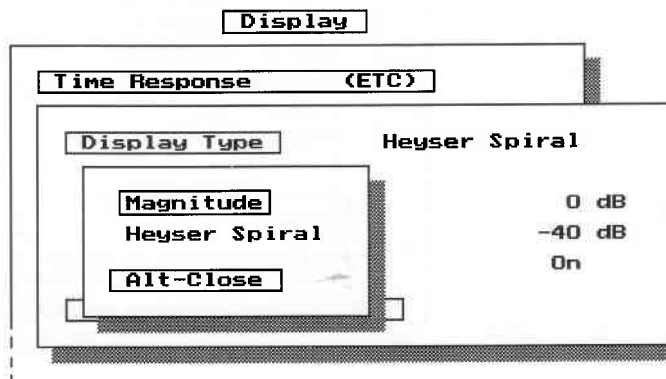


Illustration 6-5
Display parameters for the Time Response test

Running the Time Response test

► We are now ready to perform the Time Response measurement. Open the *Measure* menu and choose *Do Time Test*. You will hear the sweep last for approximately one second, and will see the ETC display appear on the screen.

You should get a display that looks similar to Illustration 6-6, depending on the similarity of your physical room setup to ours and the levels. Press *F2* to activate the data cursor.

Note: Excessive drive is not necessary to get valid data. Set the volume to a comfortable level.

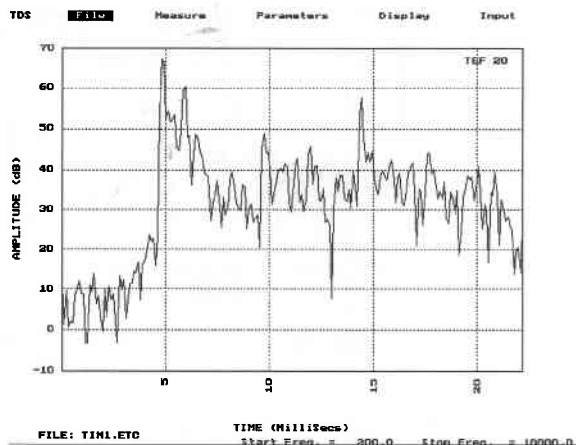


Illustration 6-6

The direct sound is the highest peak in this ETC.

The power of the data cursor

Sound Lab TDS forms a data cursor with horizontal and vertical lines that extend across the data window. The data cursor displays the value of the data at the intersection of the lines in an information box in the margin of the data window as shown in Illustration 6-7.

- Turn the *Cursor On* in the *Display* menu.

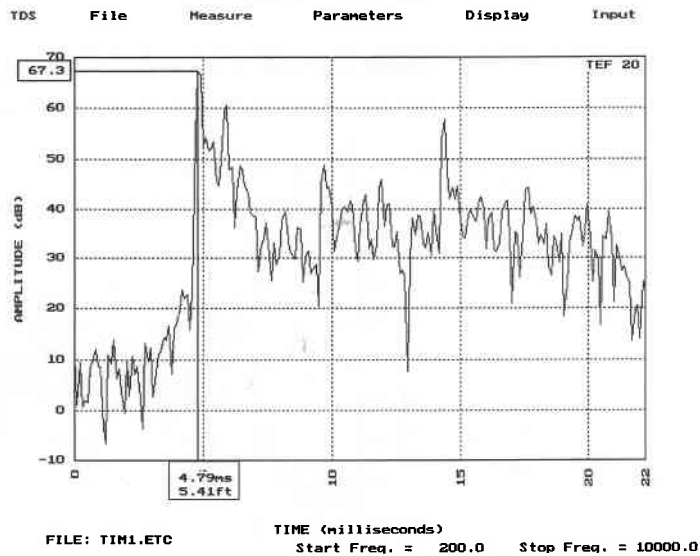


Illustration 6-7
The data cursor

Using the cursor to examine the data

Position the cursor exactly on the first large spike in the display, as in Illustration 6-7. This point represents the direct sound arrival at the microphone. The information boxes at the edge of each coordinate of the cursor give important information for evaluation of the data.

Recall that your numbers may vary from this somewhat, depending on the similarity of your testing setup to ours. The vertical coordinate information box in our test displays the dB of the amplitude of the direct sound at 67.3 dB. The horizontal coordinate information box shows the distance this sound traveled (5.41 ft) and the exact time in milliseconds that it took to travel from the loudspeaker to the microphone (4.79 milliseconds).

Note: This number (4.79 milliseconds) represents the *receive delay* that you will enter in the parameters to set up for the Frequency Response measurement we will do next.

By using the powerful cursor in *Sound Lab TDS*, you can enter the receive delay directly into the *Parameters—Frequency Response* menu. To do this, place the cursor on the peak of the direct sound and press *F4*. The receive delay is now entered into the *Parameters—Frequency Response* menu for the Frequency response test.

Finding reflections

Move the cursor over to the second peak to the right of the direct sound. The information box tells us that

at 5.91 milliseconds after the test signal, energy at an amplitude of 60.4 dB arrived, 6.68 feet out. If you subtract 5.41 feet (direct sound) from 6.68 feet, you see that there is a reflecting surface about 1.2 feet beyond the microphone. Looking at our set up, the hard surface a little over a foot away would likely be the table top. Another way

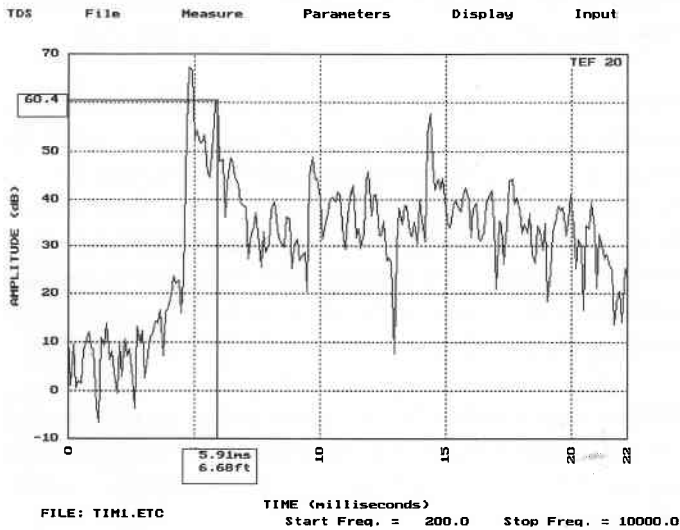
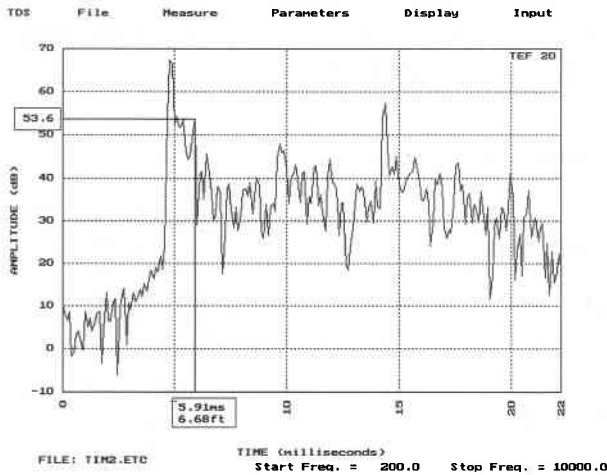


Illustration 6-8
Finding reflections

to do this would be to use the *relative cursor*. Place the cursor on the direct sound and choose the *Relative cursor button* at the lower left hand corner of the display by pressing *R* or clicking on the button with the mouse. Then move the cursor to the second peak and read the relative values in the cursor data window.

We can confirm this by placing absorptive material on the table top between the speaker and microphone; press *F5* to repeat the test.

You should note a drop in level of the second arrival. Our test, as shown in Illustration 6-9, showed the new level to be 53.6 dB, verifying that the peak was caused by a reflection off the table top.



*Illustration 6-9
Measurement
performed with sound
absorptive material on
the reflection.*

the table top. Therefore, it appears that they are caused by reflections inside the speaker box.

► Press *F2* a second time to turn the cursor off. We are now ready to do a Frequency Response test.

Some other things we see in this display are two small peaks—one 56.3 dB high at 5.78 feet, and a 53.3 dB peak at 6.13 feet. By looking at our physical set up again, there is nothing closer than

Doing a Frequency Response Test (TDS)

The Frequency response test shows how the output of a device is related to a frequency range of interest for a given amount of time. It is displayed on a graph with frequency displayed on the horizontal axis; magnitude and phase on the vertical axis.

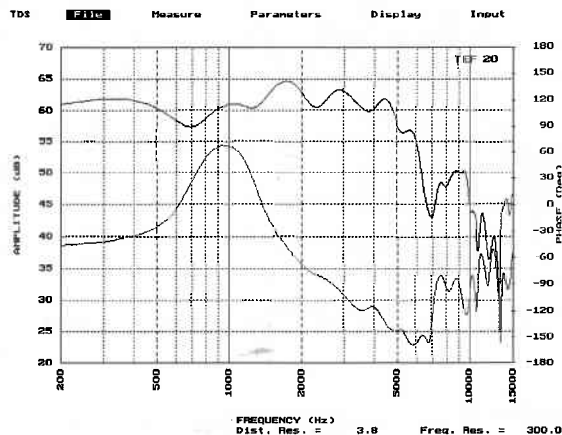


Illustration 6-10

A typical, logarithmic display of frequency/phase data.

Use the Frequency Response (TDS) measurement to:

- find comb filters,
- examine the direct sound frequency response,
- set crossover points in speaker systems,
- verify manufacturer's claims
- equalize the sound system
- confirm polarity of microphones and speakers,
- and to measure impedance over a wide frequency span.

As before, we will send a sweeping test signal into the loudspeaker and examine the data that is returned to the analyzer. When we make frequency response measurements with the TEF, we set the tracking filter of the analyzer to listen at the right time. See *Appendix A: How the TEF works*. This “time” is the time offset, or receive delay, we confirmed in the Time Response (ETC) test just performed.

Setting the parameters for the Frequency Response Test

► Open the *Parameters—Frequency Response* menu and look at the parameters. You will see that the *Receive Delay* of 4.91 milliseconds (or whatever number that you measured for your particular setup) is already entered (when we pressed *F4* in the *Time Response* display with the cursor on the direct sound).

The first decision we need to make is which frequencies we’re interested in examining, and we will set the *Start* and *Stop Frequencies* accordingly. Because of the small size of the room, we cannot set the *Start Frequency* too low because of the correlation between wavelengths and the size of the space. Recall that from the *Time Response (ETC)* display, we saw that there was 1.2 feet of space before the first reflection appeared, so we need to set the *Distance Resolution* small enough to keep the reflection out of the measurement.

- Enter the following settings in the *Parameters—Frequency Response* menu.

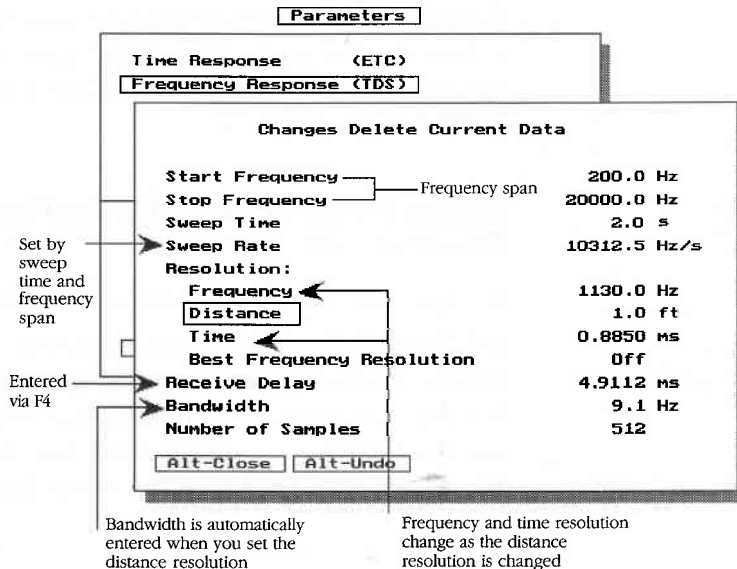


Illustration 6-11
Setting parameters for the Frequency Response test.

We want to measure within a space that will exclude the first reflection that occurred 1.2 feet from the direct sound (verified in the ETC). We will enter 1 foot *Distance resolution* to attenuate everything over 1 foot from our measurement. Notice how this entry automatically changes the *Frequency* and *Time Resolution* to correlate, as these parameters are interrelated. It also changes the bandwidth to 9.1 Hz. Setting the analyzer in this way will cause the TEF to attenuate the reflection by 15 dB.

Note: You rarely have to set the *Sweep Rate* and *Bandwidth* to do a frequency response. They are automatically set for you when you select the *Sweep Time* and *Resolutions*.

Setting the screen display

When we measure frequency with the TEF analyzer, both magnitude and phase data are recorded during the frequency sweep. Magnitude and phase are two different ways of looking at the same data. The data from a Frequency Response measurement can be displayed showing phase, magnitude, magnitude and phase, the Nyquist, or Heyser Spiral, depending on the settings you choose in the *Display* menu.

Open the *Display* menu, and make the following settings:

Display

Time Response (ETC)	
Frequency Response (TDS)	
Display Type	Magnitude
Magnitude	
Top of Scale	80 dB
Bottom of Scale	50 dB
Auto Scaling	On
Phase	
Top of Scale	180 Degr
Bottom of Scale	-180 Degr
Auto Scaling	On
Wrap	Off
Nyquist	
Top of Scale	90 dB
Bottom of Scale	60 dB
Auto Scaling	On
Graph	dB
Display Rotation	0 Degr
Horizontal Scale	Linear
Octave Smoothing	0.0 %
Alt-Close	Alt-Undo

Illustration 6-12
Display—Frequency Response parameters

Running the test

➤ Close the *Display* menu; open the *Measure* menu and choose *Do Frequency Test*. Again, you will hear the sweep last for approximately two seconds. (It will sound different this time because we are sweeping over a different frequency band), and the display will appear. It should look similar to Illustration 6-13, depending on how similar your physical setup is to ours. Press *F2* to read the data with the cursor and determine the magnitude and phase signatures.

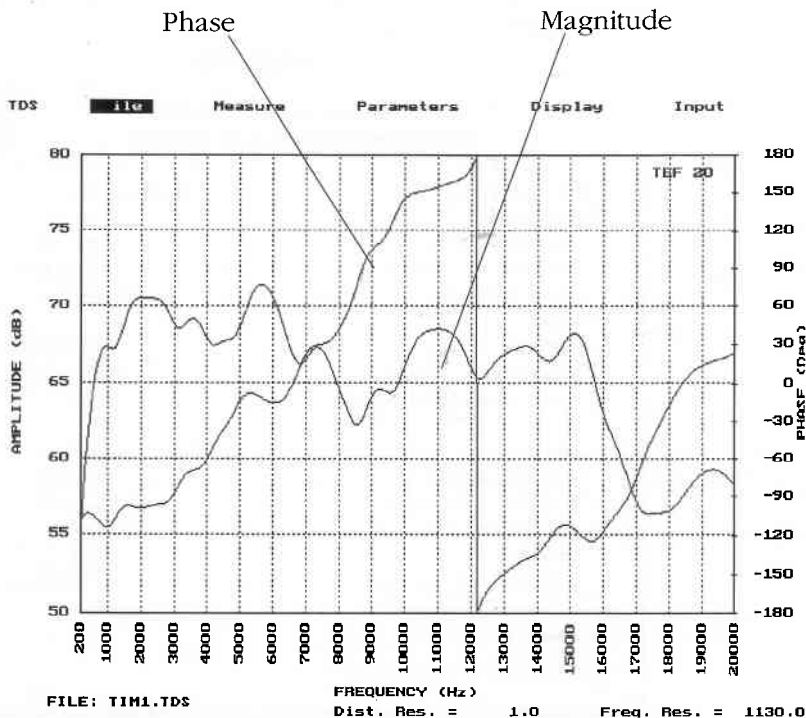


Illustration 6-13
Phase/magnitude data display.

The vertical scaling on your display may be different since we are using auto scaling. Recall we set up for a *Linear* scale, which shows most of the detail in the high end of the frequency response where it occurs. The *log* scale, however, compresses the high end, obscuring this detail. With a *linear* scale, certain problems, such as comb filtering, are more evident.

Adjusting the receive delay to tune in to the phase

The phase display is a most sensitive time measurement (remember that phase is both time and frequency dependent). When measuring phase, the correct *Receive Delay* is critical. We see that the phase display slopes upward from left to right. This slope is the signature of excessive receive delay.

Although in the previous test we had set the filter to compensate for this delay with the ETC cursor, that setting is only near the correct delay. *Sound Lab TDS* allows you to use phase data for precise adjustment of the receive delay—an accurate way to determine if you are aligned to the acoustic origin of a speaker.

A correct delay would be shown in the phase signature as relatively flat in those areas where the magnitude curve is relatively flat. A slope to the left or right would indicate that the delay is imprecisely set. You could reopen the *Parameters—Frequency Response* menu and manually experiment with different receive delays to get the desired results. *Sound Lab TDS* offers a quick way to do this.

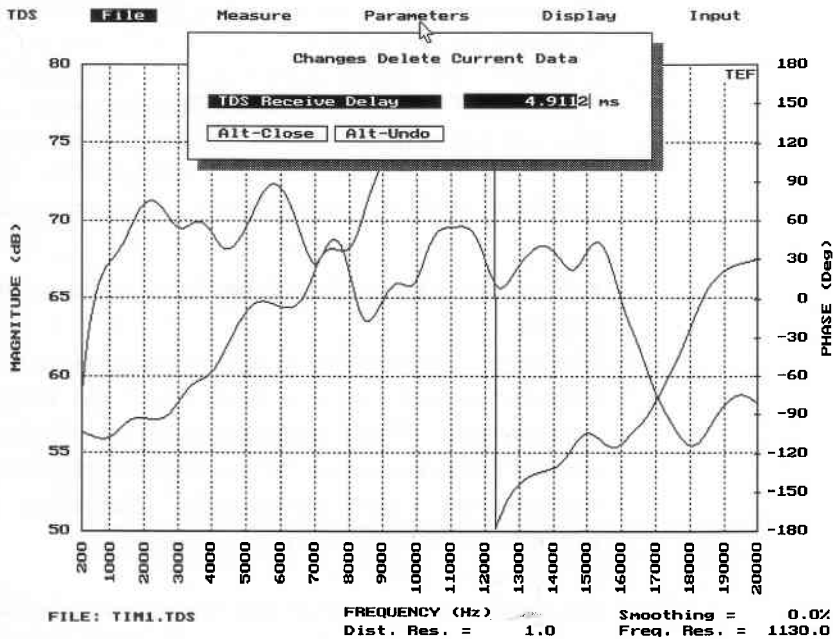


Illustration 6-14

Pressing F4 function key enables a dialog box that allow precision incremental adjustments to the receive delay.



When you press *F4* a data dialog box will appear at the top of the display. The present setting for the *Receive Delay* is highlighted and can be edited to several places from the keyboard. Use the Left and Right arrow keys to move to the place value you want to edit. You simply type in the delay that you want and then press *Enter* to have your selection entered into the parameters. You can then re-test and repeat the above procedure until the desired display is accomplished.

For most purposes, adjustments by tenths or hundredths of a millisecond are effective. The following series of illustrations shows the progression of adjustments to arrive at a precise receive delay.

Minor errors in the setting of the receive delay have virtually no effect on magnitude measurements. However, as we shall see, phase measurements are extremely sensitive and the receive delay must be properly set. For many devices, a two-way loudspeaker for example, there is no one correct receive delay for phase measurements. Unless the microphone is *precisely* the same distance from the acoustic origins of both drivers, the signals from each of the two drivers will arrive at the microphone at slightly different times, requiring two different receive delay settings for each of the drivers individually.

Illustration 6-15

Frequency/phase response data with the receive delay entered directly from the ETC at 4.91 milliseconds. The upward slope of the phase data indicates the delay is set past the correct time.

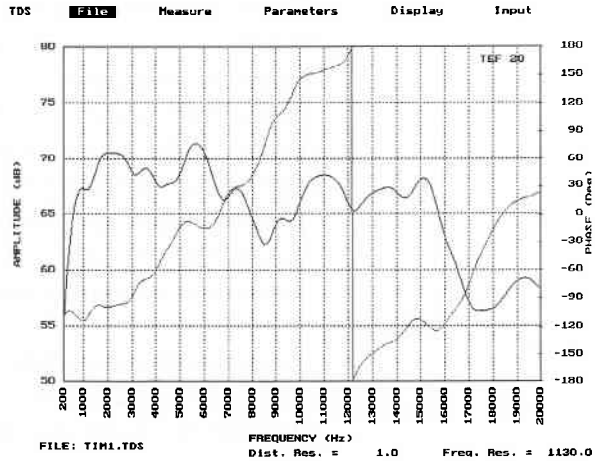


Illustration 6-16

Frequency/phase response data with the receive delay set 0.1 millisecond earlier. This information was entered into the parameters from the information box. The downward slope of the phase data indicates that the setting was too far before the correct time.

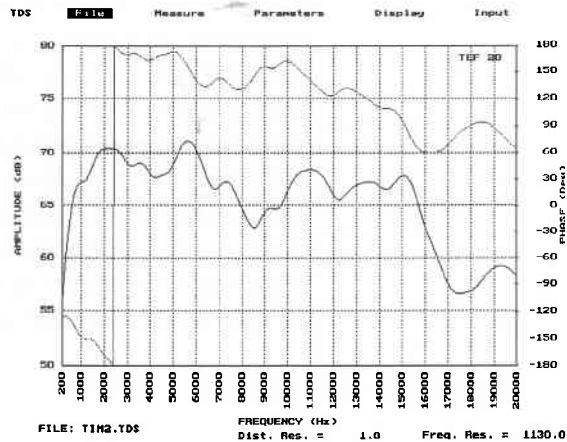


Illustration 6-17
Frequency/phase response
data with receive delay
set 0.05 millisecond later
than Illustration 6-14.

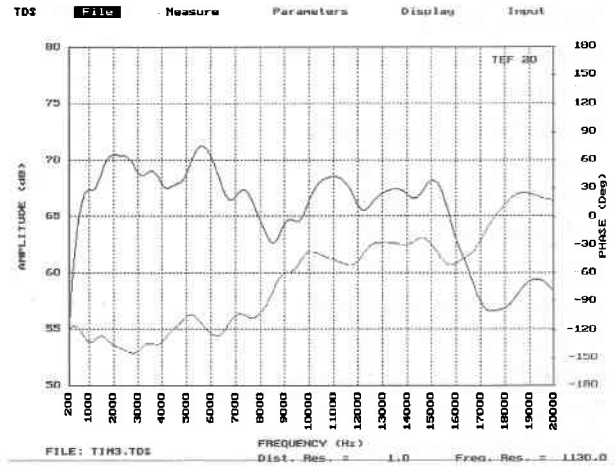


Illustration 6-18
A 0.01 millisecond
adjustment to
measurement in
Illustration 6-15 creates
an even flatter phase
response.

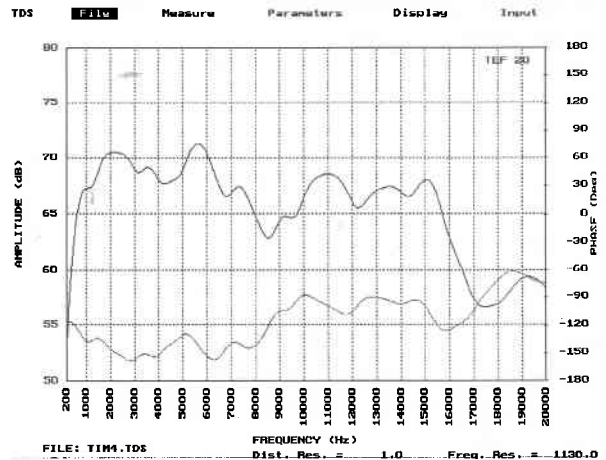
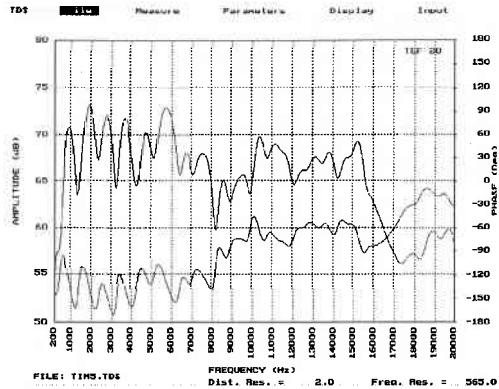


Illustration 6-19

In this illustration, we returned to the Parameters—Frequency response menu, and changed the distance resolution to actually include the reflection that was identified at the first peak of the ETC test we performed. Setting the Resolution—distance to 2 feet, and re-running the test caused this display, showing the comb filtering produced by the reflection's presence in the measurement.



Making %ALcons and RT60 measurements

For the purposes of this exercise, we are assuming the user to be a contractor or designer with a fundamental understanding of navigation in *Sound Lab TDS* software. The task is to calculate %ALCONS and RT60 data out of a valid ETC measurement in order to

- meet a bid specification
- verify the reverberation time of the room
- determine if the decay is appropriate for the functions that are going to occur in this environment.

Sound Lab TDS performs two common acoustic calculations useful to the designer, consultant, and contractor—the RT60 and %ALCONS. Both are calculated from data collected in a Time Response (ETC) measurement. It follows, then, that to yield valid RT60 and %ALCONS numbers, you need to have appropriate ETC data.

Terms to know

To have a working knowledge of the exercise to follow, here are terms to know:

%ALCONS—The measured percentage of Articulation Loss of Consonants by a listener. With TEF methods, articulation scores are measured as percent of articulation loss of consonants in speech. A %ALCONS of 0 indicates perfect clarity and intelligibility with no loss of consonant understanding, while 10% and beyond is growing toward bad intelligibility, and 15% typically is considered the maximum allowable.

Early Decay Time—The time for a reverberant sound field to decay 10 dB below the level of the direct sound. Short decay times cause music and speech to sound dry or muffled. Long decay times make speech unintelligible and difficult to understand. It is the figure that most closely approximates how the decay time “sounds” to the ear.

RT60 (Reverberation time)—The time in seconds for the reverberant sound field to decay 60 dB after the sound source is shut off. It is calculated by measuring the rate of decay over as much decay as possible in the curve (ideally, the first 25 dB to 30 dB of decay) and extrapolating what the RT60 would be if the decay continued at that rate.

An appropriate ETC

RT60 and %ALCONS calculations begin with an appropriate ETC measurement—either an individual measurement or one taken as a part of an STI data set.

There are three fundamental requirements to have an ETC that is valid to calculate RT60 and %ALCONS.

- The room must be large enough to have a statistical reverberant field.
- The ETC display on the TEF analyzer must be of sufficient duration to see the reverberant field.
- The sweep must be slow enough to excite the room and allow the reverberant sound to return to the TEF analyzer. To be certain you sweep slowly enough, you should set the sweep time to be 3-4 times longer than the time you want displayed on the screen.

ETC display example

To see an example of an appropriate ETC from which to calculate RT60 and %ALCONS, open the data file called SNCTONLY. ETC from the data which accompanied your Sound Lab TDS software. This display is a Time Response (ETC) showing the first second of decay in a room with the sweep centered at 2000Hz.

This ETC display is representative of a reasonably well-behaved 200 seat room. A general look at the display without the cursors indicates coherent, direct sound, approximately 8 dB in level above the nearest reflection. We see a reverberation time that is less than one

second, with no significant arrivals later in time. Just by looking at the time response raw data, the room appears to be very good. The %ALCONS calculation we will do takes all the above considerations, and puts them into one number.

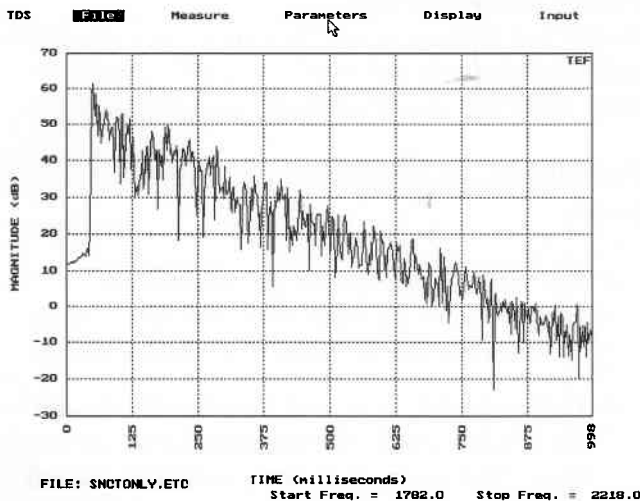


Illustration 6-20 Time Response (ETC)

Parameters

If you note the parameters used to set up for this display, you will see that the start frequency of this test is 1782 and the stop frequency is 2218. This gives a center frequency of 2000. The 2 kHz octave band contains approximately 1/3 of all intelligibility in speech. Since we are examining speech, these settings will center the information around 2 kHz, a typical starting point for the first sweep you would do.

The time span of approximately one second was estimated by "testing" the room with a simple handclap, and listening to how long it took the sound to decay.

Power of the cursor

Sound Lab TDS has powerful cursors you can activate that automatically analyze and report RT60 and %ALCONS data. Make the cursor active by pressing *Shift F2*.

You will now see the following new information on the screen.

- The smooth line above the original data is an integration of the time response data, which smooths the decay curve. (Schroeder curve)
- Three cursors, *L*, *D*, and *R* across the bottom of the display, each with accompanying information. The *L* and *R* cursors represent the left and right points of the reverberant field that will be used for determining the early decay time.

- The third cursor, labeled *D*, sets the division between early (direct sound) and late (reverberant field) energy.¹ *Sound Lab TDS* software uses this information to compute the Direct to Reverberant Energy Ratio.
- Cursor buttons at the bottom of the display.

Slope displays the line which best fits the data between the left and right cursors. (Linear regression line.)

TDS Delay automatically enters the value at the point of the active cursor into the the *Receive Delay* option of the Parameters for the Frequency Response test.

3D Start Delay button automatically enters the value at the point of the active cursor into the *Start Receive Delay* option of the Parameters for the 3D test.

3D End Delay button automatically enters the value at the point of the active cursor into the *End Receive Delay* option of the Parameters for the 3D test.

Additional information

You will also see additional information in boxes across the top of the display which represent the software's best estimate for placing the cursors.

- RT60 = 0.62 Sec (reverberation time or early decay time)
- EDir/ERev = -2.5 dB (early direct-to-early reverberant sound ratio)
- ALCONS = 3.81% (the %ALCONS),
- dB down = 10.5 (the difference in level on the integration line between the left and right cursors.)

These initial cursor positions and values in the information boxes represent the computer's best estimate regarding placement. Inspect the display carefully and make adjustments as necessary.

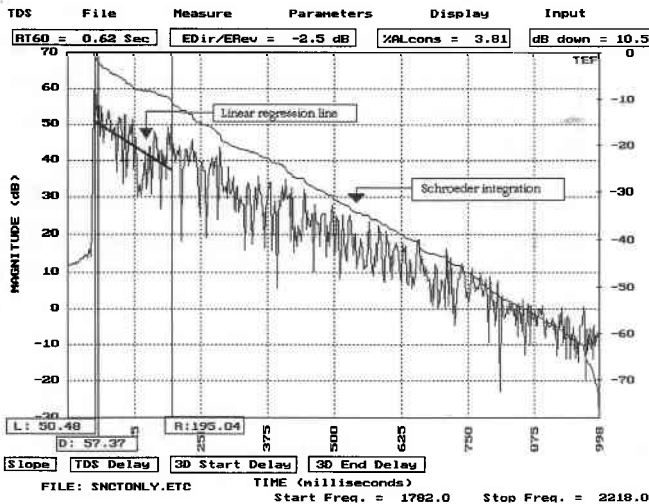


Illustration 6-21

Data cursors analyze and report RT60 and % ALCONS data.

Classical RT60

You can manipulate the cursors on the analyzer to accomplish a more classical RT60 value by manually placing the cursors on the display such that the left and right cursors span the longest and smoothest linear range possible. This includes as much dynamic range as possible in the calculation.

To move the cursors manually, you must select them and move them with the mouse or arrow keys

- Place the *R* cursor to the right end of the linear range at 720.52 milliseconds.
- Place the *L* cursor to the starting point of the linear portion of the decay on the ETC, at 192.75 milliseconds.
- The reverberation time shown in the RT60 information box now reads 0.92 seconds.² See illustration on next page.

Note: At any point when the RT60 cursors are active, you can see the actual linear regression line by choosing the *Slope button* or pressing the *Quick key O*.

You can see an illustration of this on the next page.

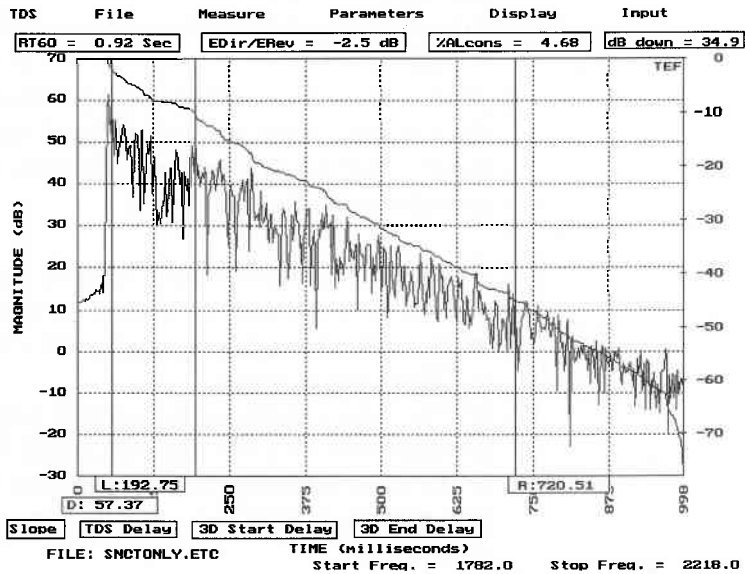


Illustration 6-22
Classical RT60 numbers calculated by the TEF analyzer.

Early Decay Time

The Early Decay Time is a number that corresponds to perceived decay time.

- To find the Early Decay Time, put the left cursor on the direct sound at 50.48.
- Move the right cursor until the dB down information box shows as close as possible to 10 dB (9.8). The RT60 information box at the top of the screen will now indicate the Early Decay Time (RT60) of 0.58 seconds.

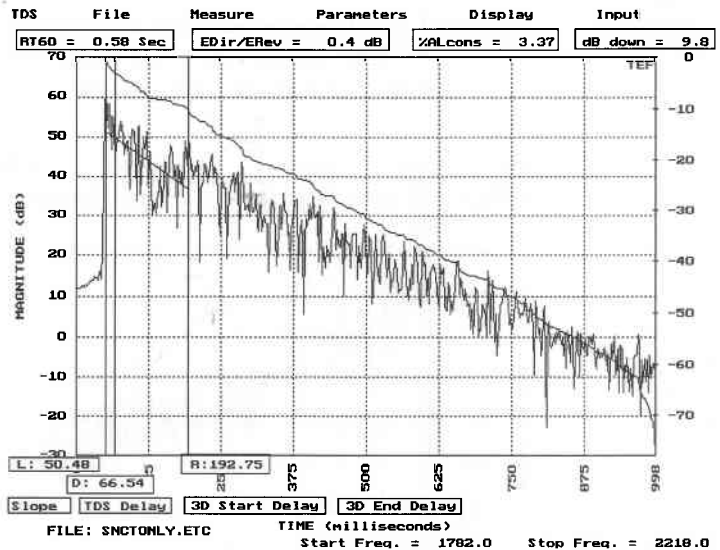


Illustration 6-23

Using the cursor to find the early decay time.

Finding the %ALCONS

- To find the %ALCONS, place the left and right cursors as for the Early Decay Time. Look at the first 20 milliseconds after the direct sound for the presence of any significant reflections. Since such reflections will be perceived as direct sound, you must include them in the direct sound portion of the calculation. To do this
 - Place the *D* cursor on 66.54, one point to the right of the reflections
 - The %ALCONS in the %ALCONS information box reads 3.37.

If no strong reflections are present,

- Place the *D* cursor one point to the right of the direct sound and read the %ALCONS.

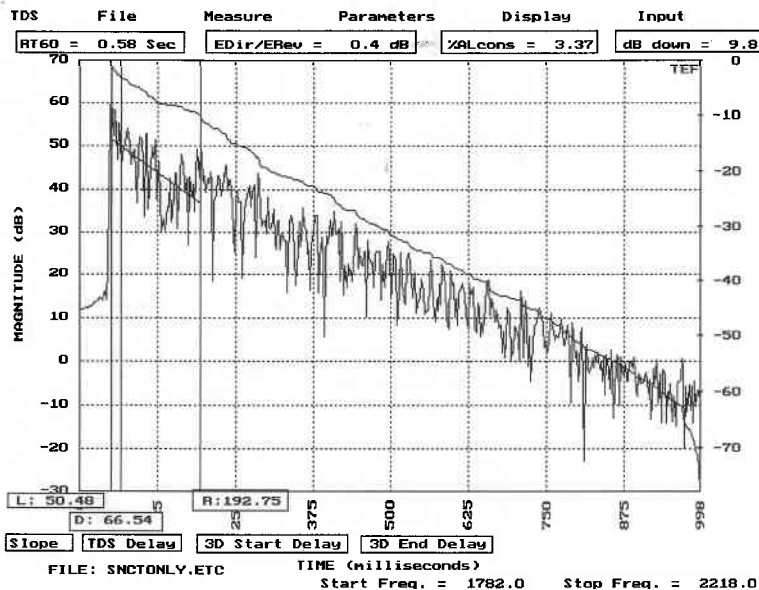


Illustration 6-24
Finding the %ALCONS

Appendix A: Basics of sound

Waves

To produce sound, something vibrates against air molecules which pick up the vibration and pass it along as sound waves. When these vibrations strike our ears, we hear sound.

Let's examine how sound waves are created. Suppose a speaker cone in a guitar amp is vibrating — moving rapidly in and out. When the cone moves out, it pushes the adjacent air molecules closer together. This forms

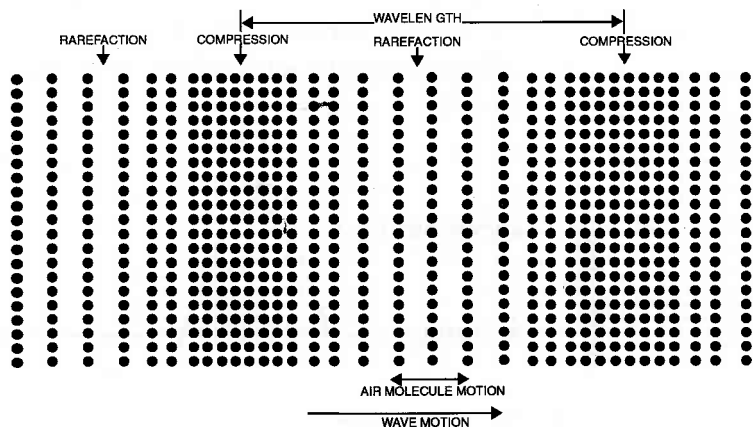


Illustration A-1 A sound wave is made of high-pressure compressions and low-pressure rarefactions.

a compression. When the cone moves in, it pulls the molecules farther apart, forming a rarefaction. As illustrated in Illustration A-1, the compressions have a

higher pressure than ambient atmospheric pressure; the rarefactions have a lower atmospheric pressure than normal.

These disturbances are passed from one molecule to the next in a spring-like motion to pass the wave along. The sound waves travel outward from the sound source at about 1130 feet per second.

At some receiving point, such as an ear or a microphone, the air pressure varies up and down as the disturbance passes by. Illustration A-2 is a graph showing how sound pressure varies with time. It fluctuates up and down like a wave; hence the term "sound wave." The high point of the graph is called a peak; the low point is called a trough. The horizontal center line of the graph is normal atmospheric pressure.

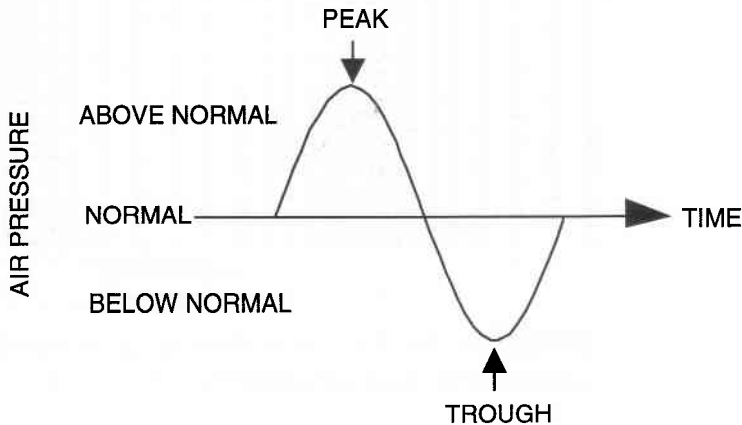


Illustration A-2

Sound pressure vs. time of one cycle of a sound wave.

Characteristics of sound waves

Illustration A-3 shows three successive cycles of a wave. One complete vibration from high to low pressure and back to the starting point is called one cycle. The time between the peak of one wave and the peak of the next is called the period of the wave. One cycle is one period in time long.

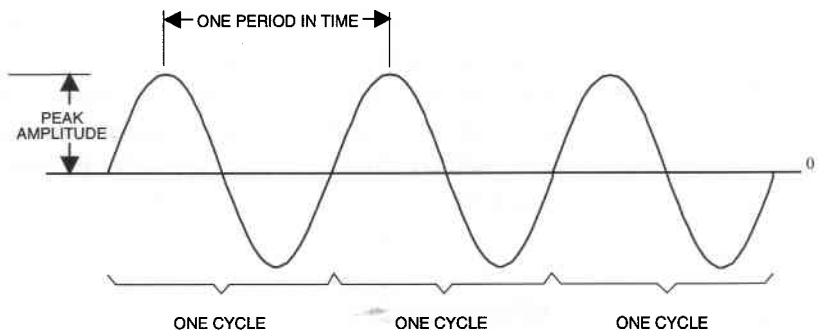


Illustration A-3 Three cycles of a wave.

Amplitude

At any point on the wave, the vertical distance of the wave from the center line is called the amplitude of the wave. The amplitude of the peak is called the peak amplitude. The more intense the vibration, the greater the pressure variations, and the greater the peak amplitude. The greater the amplitude, the louder the sound.

Frequency

The sound source (in this case, the loudspeaker) vibrates back and forth many times a second. The number of cycles completed in one second is called the frequency. The faster the speaker vibrates, the higher the frequency of the sound. Frequency is measured in hertz (abbreviated Hz., One Hertz equals one cycle per second).

The higher the frequency, the higher the perceived pitch of the sound. Low-frequency tones (say, 100 Hz) are low pitched; high-frequency tones (say, 10,000 Hz) are high- pitched. Doubling the frequency raises the pitch one octave.

Wavelength

When a sound wave travels through the air, the physical distance from one peak (compression) to the next is called a wavelength. (This was shown in Illustration A-3.) Low frequencies have long wavelengths (several feet); high frequencies have short wavelengths (a few inches or less).

Phase and phase shift

The phase of any point on the wave is its degree of progression in the cycle — the beginning, the peak, the trough, or anywhere in between. Phase is measured in degrees, with 360 degrees being one complete cycle. The beginning of a wave is 0 degrees; the first peak is 90 degrees ($1/4$ cycle), and the end is 360 degrees.

Illustration A-4 shows the phase of various points on the wave.

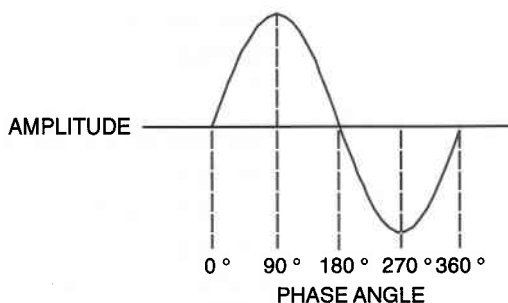


Illustration A-4

The phase of various points on a wave.

If there are two identical waves, but one is delayed with respect to the other, there is a phase shift between the two waves. The more delay, the more phase shift. Phase shift is measured in degrees. Illustration A-5 shows two waves separated by 90 degrees (1/4 cycle) of phase shift.

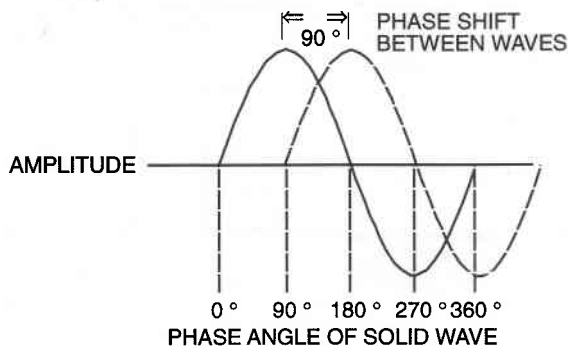


Illustration A-5

Two waves 90 degrees out-of-phase. The dashed wave lags the solid wave by 90 degrees.



When there is a 180-degree phase shift between two identical waves, the peak of one wave coincides with the trough of another. If these two waves are combined, they cancel each other out. This phenomenon is called phase cancellation.

Harmonic content

The type of wave shown in Illustrations A-2 and A-5 are called sine waves. A sine wave is a pure tone of a single frequency, such as produced by a tone generator. However, most musical tones have a complex waveform, which has more than one frequency component. Yet no matter how complex, all sounds are combinations of sine waves of different frequencies and amplitudes.

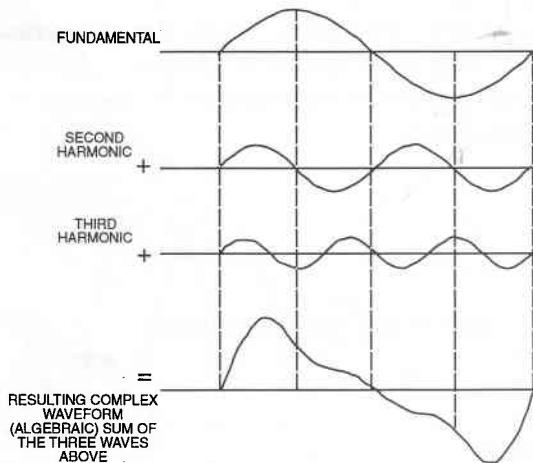


Illustration A-6 shows sine waves of three frequencies combined to form a complex wave. The amplitudes of the various waves are added algebraically at the same point in time to obtain the final complex waveform.

Illustration A-6 Addition of fundamental and harmonics to form a complex waveform.



The lowest frequency in a complex wave is called the fundamental frequency. It determines the pitch of the sound. Higher frequencies in the complex wave are called overtones or upper partials. If the overtones are integral multiples of the fundamental frequency, they are called harmonics. For example, if the fundamental frequency is 200 Hz, the second harmonic is 400 Hz (2×200); the third harmonic is 600 Hz (3×200), and so on.

The number of harmonics and their amplitudes relative to the fundamental partly determine the tone quality or timbre of a sound. They identify the sound as being from a trumpet, piano, organ, voice, etc. White and pink noise contains all audible frequencies and has an irregular, non-periodic waveform.

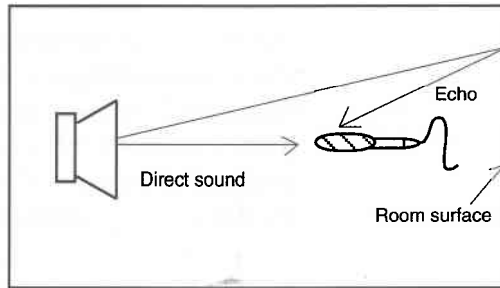
Behavior of sound in rooms

So far we've covered the characteristics of sound waves traveling in open space. But since most music is heard in rooms, we need to understand the acoustic phenomena created by the room interior surfaces.

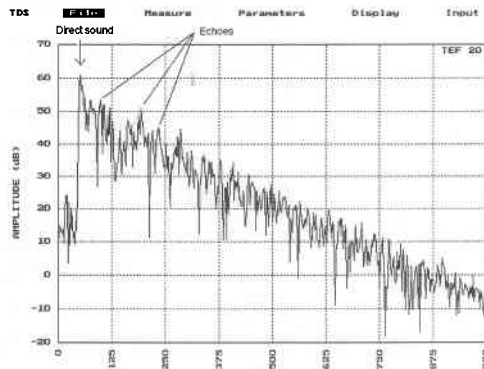
Echoes

A sound source vibrates against air molecules, creating sound waves that travel outward in all directions. Some of the sound travels directly to the listener (or to a microphone) and is called direct sound. The rest strikes the walls, ceiling, floor, and furnishings of the recording room. At those surfaces, some of the sound energy is absorbed, some is transmitted through the surface, and the rest is reflected back into the room.

Since sound waves travel about 1 foot per millisecond, the sound reflections arrive after the direct sound reaches the listener. The delayed arrival of a reflected sound causes a repetition of the original sound called an echo, as shown in Illustration A-7). In large rooms we sometimes hear discrete single echoes; in small rooms we often hear a short, rapid succession of echoes, evenly spaced in time, called flutter echoes.



A



B

*Illustration A-7 Echo (A) Echo formation.
(B) Amplitude vs. time of direct sound and echo.*



Parallel walls or diagonally opposite corners create flutter echoes by reflecting sound back and forth between them many times. You can detect flutter echoes by clapping your hands next to one wall and listening for a fluttering sound. Since echoes can reduce the clarity of sound, they should be eliminated by adding patches of absorbent material or diffusors to one or both of the offending walls. Putting the material in patches, rather than all together, promotes an even distribution (diffusion) of sound in the room.

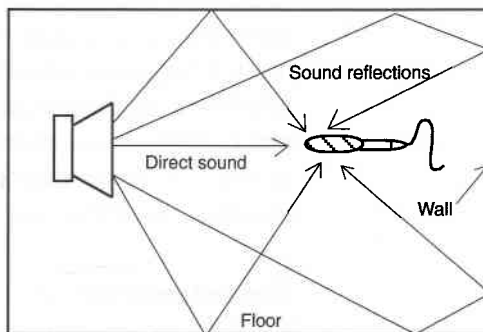
Reverberation

Sound reflects not just once but many times from all the surfaces in the room. These sonic reflections sustain the sound of the instrument in the room for a short time even after the sound source is stopped. This phenomenon is called reverberation — the persistence of sound in a room after the original sound has ceased. For example, reverberation is the sound you hear just after you shout in an empty gymnasium. The sound of your shout persists in the room and gradually dies away (decays).

In physical terms, reverberation is a series of multiple echoes, decreasing in intensity with time, so closely spaced in time as to merge into a single continuous sound, eventually being completely absorbed by the inner surfaces of a room. Echoes increase in number as they decay. Illustration A-8 shows reverberation as a decay-in-time of room reflections.

Note that reverberation is a continuous fade-out of sound, while an echo is a discrete repetition of a sound.

A



B

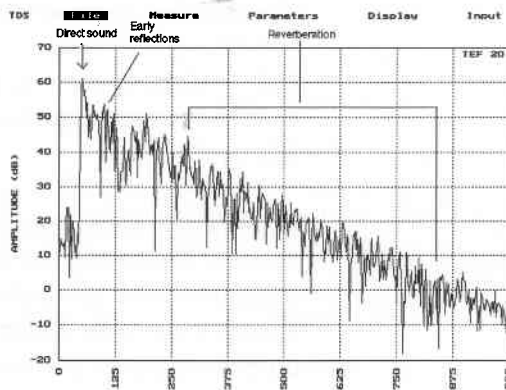


Illustration A-8 Reverberation. (A) Reverberation formation. (B) Amplitude vs. time of direct sound, early reflections, and reverberation



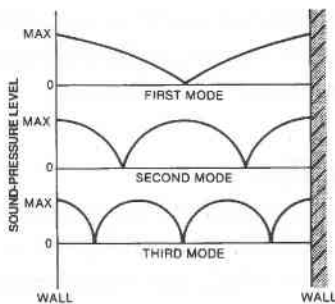
Reverberation time

The time it takes for sound to decay to 60 dB below the original steady-state sound level is called reverberation time (abbreviated T or RT60).

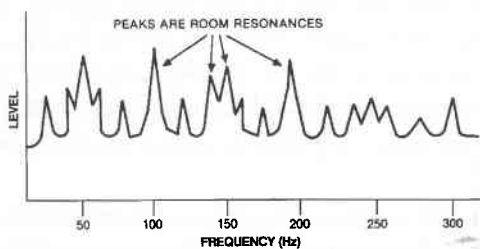
Room modes

If you play an amplified bass guitar through a speaker in a room, and do a bass run up the scale, you will hear some notes at which the room resonates, reinforcing the sound. These resonant frequencies, most noticeable below 300 Hz, are called room modes or normal modes. Resonance peaks of up to 10 dB can occur. They give a tubby or boomy coloration to musical instruments and should be minimized. Room modes occur in physical patterns called standing waves.

Standing waves are uneven sound-level distributions in a room caused by sound waves continuously reinforcing themselves as they reflect between opposing surfaces. Opposite walls (or the ceiling and floor) can support standing waves between them, as shown in Illustration A-9. Weaker modes can occur between other surfaces.



A.



B.

Illustration A-9 Standing-wave phenomena.

A. *Pressure distribution between two opposing walls, for first three room modes.*

B. *Example of frequency response of a room with standing waves.*



The frequencies at which the room resonates depend on the dimensions of the room — its length, width, and height. The formula for the most basic room-mode resonance frequencies is

$$f = (N \times 565)/D$$

where f = resonance frequency, in Hz

$N = 1, 2, 3, \dots$

D = room dimension, in feet.

For example, a room 12 feet long will have room modes at 47 Hz, 94 Hz, and so on. Those frequencies or notes will be over-emphasized in the music unless there is sufficient bass trapping in the room to dissipate them.

Other frequencies will be reinforced by other room dimensions. If the height, width, and length of the room are identical, the same modal frequencies will be reinforced in all three dimensions, greatly emphasizing certain low frequencies. On the other hand, if the dimensions are not multiples of each other, the modes will be different for each dimension. Then, each room mode will be reinforced in only one dimension and there will be a more even distribution of resonance frequencies.

Appendix B: How TEF measurements work

The TEF analyzer generates a sine-wave frequency sweep which is played through a sound system and returned to the TEF. The change in frequency of the sweep is linear with time. The microphone signal is fed through a filter that tracks the sweep. The tracking filter is in sync with the generated frequency sweep; however, it is offset in time to compensate for the propagation delay of sound traveling from speaker to microphone. By varying the bandwidth and time offset of the tracking filter, you can study the spectrum of the direct sound by itself, certain reflections, or both.

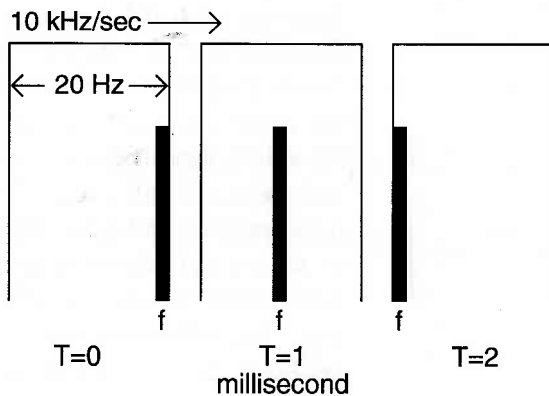


Illustration B-1
A filter sweeping over frequency of interest.

For example: A 20 Hz-wide filter sweeps along at 10,000 Hz/second. (For clarity, the filter is drawn with a rectangular shape. The edges of the rectangular filter correspond to the 3 dB down points of a real filter.) At the left edge, the frequency of interest (f) is just entering the filter's bandwidth. At the center of the Illustration B-1, the filter has swept higher in frequency and is tuned to the frequency of interest. At the right, the frequency of interest is leaving the filter—the filter having swept still higher. Quite simply, the TEF puts an exactly measured signal into a system, it knows exactly when to listen for it to emerge, and knows exactly how to figure out what these results mean.

Reflections can be kept out of measurement

Suppose a 1000 Hz tone is swept to the microphone through the air. At that instant, the tracking-filter center frequency is set to 1000 Hz. Now suppose that the loudspeaker's sound reflects off a wall and enters the microphone after a certain delay. By the time the reflection enters the microphone, the tracking filter will have swept to a higher frequency than the reflection, as shown in Illustration B-2. If the filter bandwidth is sufficiently narrow, the reflection is rejected or filtered out. No reflection signals are received by the TDS analyzer. In other words, an anechoic measurement is made in an ordinary room.

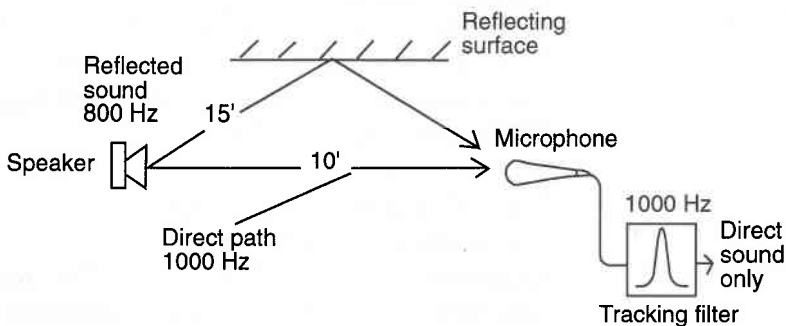


Illustration B- 2

A reflection is filtered out of a TEF measurement.

Relationship between Time/Frequency

Any method used to measure frequency requires a time interval in which to measure it. For example, if we want to describe the motion of a pendulum, we might say that the pendulum swings from right to left and back again in $1/2$ second. That is, it has a period (of time) of $1/2$ second. This is a description in the time domain. Alternatively, we might say that the pendulum moves from right to left and back again with a frequency of 2 Hz (2 cycles per second). This is a description in the frequency domain. Both are correct, and each is required to measure the other. A time interval is required to measure frequency regardless of the measuring method used.



Fourier transform

Frequency information is mapped into time information by means of a calculation called the Fourier transform. The Fourier transform relates time and frequency for TEF's sweeping oscillator.

The unit of frequency is defined as the reciprocal of the unit of time; then mathematically, the unit of time is the reciprocal of the unit of frequency. The mathematical descriptions of this relationship between time and frequency are:

$$\text{Time} = 1/\text{Frequency} \quad (T = 1/F)$$

where T is the period in seconds and F is the frequency in hertz, and

$$\text{Frequency} = 1/\text{Time} \quad (F=1/T)$$

where F is the frequency in hertz and T is the period in seconds.

The product of time (period) versus frequency will always equal one;

$$\text{Time} \times \text{Frequency} = 1 \quad (T \times F = 1)$$

For example, if we have a 20 Hz sine wave, we can calculate its period as being 1/20 or .05 second. If we measure the period of a cyclic process to be .1 second, then its frequency is 1/0.1 or 10 Hz. This reciprocal relationship is always present and needs to be considered when setting parameters.

Measurement resolution

When any quantity is measured, it is always limited in its resolution, or how much detail we can see, or resolve. Resolution is the degree of clarity with which we can observe or measure something. For example, you can look at a butterfly with the unaided eye and observe its coloration and veining, and you can see that it has intricate markings.

If you examine closer with a magnifying glass, you will be able to see more fine detail of its markings because you have increased the resolution of your view. If you place a butterfly wing under a microscope, you would possibly be able to see or resolve its cellular structure because you have increased the resolution dramatically. Note, that as you increase resolution looking at the wing, you know less about the total picture, i.e. how big it is, what color the body is, etc.

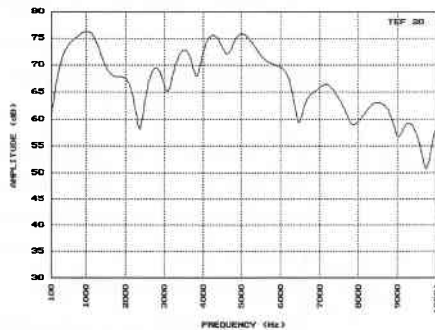
If we make a time-domain measurement with a resolution of one millisecond, then we will be unable to see any fine details that occur faster than one millisecond. We will have a frequency resolution of $1/0.001$ or 1000Hz. The details will not disappear completely, but they will be blurred or smeared, in much the same way as viewing the butterfly unaided. In order to measure 10 Hz, we require $1/10$ or .10 seconds or longer to measure one full period. In order to measure one second, we need a frequency of 1 Hz or greater.

The same is true in the frequency domain. If we make a frequency domain measurement with a resolution of 1 kHz, then we will be unable to see any fine details that



occur less than 1 kHz apart. The effect of this on a frequency response curve is to smooth it out and minimize the peaks and valleys.

A



B

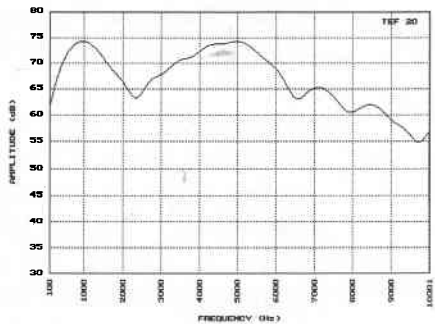


Illustration B-3

Two different resolutions of a frequency response.

Illustration B-A is a frequency response made with 500 Hz resolution. The same frequency response, made with 1 kHz resolution, is shown in Illustration B-3b. Notice the smoothing effect when measuring at lower

frequency resolution (1kHz). The fine detail is still there in reality—we just can't see it.

Trade-off between time and frequency

Because time and frequency resolutions are reciprocals of each other, we can trade away resolution in one domain for resolution in the other. However, there is always an inverse trade-off when converting from one domain to another; this means that as the resolution in one domain goes to infinity, the resolution in the other domain must approach zero. The product of resolution in time and resolution in frequency will at best equal one. With TEF it is always one.

For example, if we wish to measure a loudspeaker with a resolution in frequency of 20 Hz, then our resolution in time is $1/20$ or .05 seconds. At the speed of sound, .05 seconds is equivalent to 56.5 feet. Reflections from objects within that distance will be included in our measurement yielding false results. If we now reduce our resolution in frequency to 500 Hz, our resolution in time increases to $1/500$ or .002 seconds. This corresponds to a distance resolution of 2.26 feet and true anechoic measurements are easily performed in smaller spaces.

Note that the limits on frequency resolution are not unique to TDS, as TDS measurements perform at the theoretical limits of time-frequency resolution. The limits are a direct consequence of frequency resolution being the inverse of time resolution and applies to all measurement systems.

By deriving one domain from the other we do not learn anything new about the thing we are measuring. We have merely converted our description of it from one domain to the other. The description in either of the domains is sufficient to describe the other. The factors that should determine which domain to measure are those of convenience. If, for example, one domain permits measurements in seconds rather than minutes, then the domain providing the faster measurement should be used. Because the TEF Analyzer measures in the frequency domain, it is able to achieve significant increases in signal-to-noise ratio over equivalent time domain techniques.

The accuracy and acuity of the measurement depends entirely on the sensitivity of the measuring instrument and the resolution of its display. In general, when measuring a system, make initial measurements in wide bands of frequency or wide slices of time, then narrow these bands to increase the resolution of the measurements.

Interrelated parameters

Sweep rate

Slow sweeps can be used to provide high resolution TDS measurements in noisy environments or environments where the sweep level must be below the perceptible level of the noise.

A zero sweep rate is mathematically equivalent to a single fixed frequency. At slow rates, any amount of frequency resolution is available at the expense of time information. Conversely, high sweep rates are used to

increase time resolution at the expense of frequency resolution.

Frequency span

The second parameter to set is the frequency range of the sweep. This range should roughly correspond to the range over which useful operation of the system is to be expected. Stimulus should not be applied for long durations which might damage the system.

Sweeping through zero Hertz is generally desirable when doing ascending TDS sweeps if you want improved resolution at low frequencies. Most transducers will not be damaged by this practice if you avoid very slow sweep rates at high power levels.

Receive (signal) delay

When making measurements involving a signal delay, the third setting is the receive (signal) delay. Normally you set the receive delay equal to the travel time of sound from loudspeaker to microphone.

$$T = D/C$$

where T = receive delay in seconds,

D = distance between speaker and microphone in feet and

C = the speed of sound, 1130 feet per second.

For energy vs. time curves or ETCs, the usual practice is to use a zero receive delay between the test and analyzer oscillations. This is not necessary, if it is

known that no signal arrives at the analyzer before a certain delay. If you program this value of delay, the ETC display will start at this value of delay instead of zero.

Bandwidth/Sweep rate

When making measurements with Sound Lab, you set the desired time, distance and frequency resolutions in the Parameters menu. The TEF will then set the combination of sweep rate and bandwidth of the tracking filter accordingly. While sweep rate and bandwidth can be set manually, it is easiest to set a frequency resolution and a sweep time and let the computer automatically set the sweep rate and bandwidth for you.

There are an unlimited number of combinations of sweep rate and bandwidth that will result in a given set of resolutions. The choice should be governed by the environment noise levels because, as the bandwidth is narrowed, the probability of interference is reduced.

Slower sweep rates mean less bandwidth in the tracking filter, thus less noise getting into the measurement. With a slower sweep rate, greater total energy is put into the test, creating improved signal to noise ratio.

As bandwidth is reduced and resolution remains fixed, the time to take the measurement is increased because Sound Lab will change the sweep time to keep the parameters valid. If the testing environment is relatively free of noise, fast sweeps can be used. If noise is a problem, slow sweeps should be used.

This capability is one of TEF's greatest advantages over other systems. When there is little noise present, you can sweep very rapidly and measure from 0 Hz to 20 kHz in less than one second. If there is a high level of ambient noise, you can slow the sweep, thereby narrowing the bandwidth of the filter in order to maintain resolution and reject the noise. The noise is rejected by the narrowness of the filter. Very little of the noise signal is going to be present in a 2 Hz wide filter

Measurement repeatability should be used as a criteria to select the highest reasonable sweep rate. If the noise or sweep rate is too high, the measurement will change from one sweep to the next.

The wider the bandwidth setting, the greater the "time window," or range of time over which signals are accepted by the analyzer.

The relation between time window, bandwidth, and sweep rate is:

$$T = B/S$$

where T = width of time window in seconds

B = bandwidth in Hz and

S = sweep rate in Hz/seconds

Since sound travels a certain distance within a time interval, the time window corresponds to a "space or distance window". The space window is an ellipsoid space around the speaker and microphone, inside of which sound reflections are included in the

measurement. The speaker and microphone are at the foci of the ellipsoid. Sound reflections originating outside the space window are excluded from the measurement. Actually, they are attenuated 3 dB at the edge of the space window ellipsoid, and by greater amounts outside that.

For example, suppose you had a two millisecond time window. Since sound travels at 1130 feet/second, during a two millisecond time window, it will travel 2.26 feet. Therefore, we have a 2.26 feet distance window. This means that any signal that has a total path length that is within plus or minus 1.13 feet (1/2 of 2.26 feet) of the direct sound will be included in our measurement.

Space window considerations

On the TDS analyzer, the space window is determined by the settings of the bandwidth and sweep rate. For example, a 10-foot space window would correspond to a bandwidth setting of 88.5 Hz at a sweep rate of 10,000 Hz/second.

Here is the appropriate formula:

$D = BC/S$, where

B = bandwidth setting of tracking filter in Hz

S = sweep rate in Hz/second

D = space window in feet

C = speed of sound, 1130 feet/second

The larger the space window, the lower the frequency that can be measured accurately. That is, the lowest frequency that can be measured without interference decreases as the space window increases. Therefore a

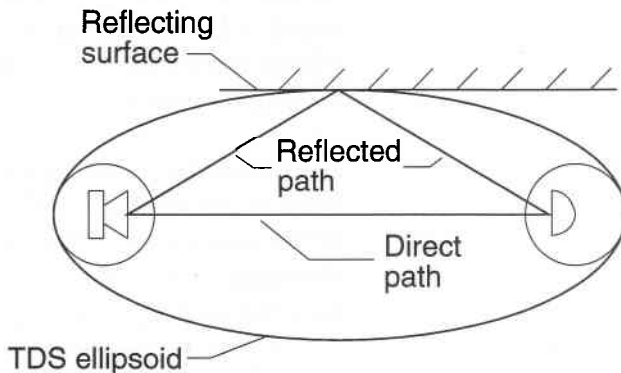


Illustration B-4
The space window, or TDS ellipsoid.

relatively large, empty room is needed for low-frequency measurements. This applies to all measurement systems.

As stated before, reflections from surfaces at the edge of the space-window ellipsoid are attenuated 3 dB. This attenuation may be inadequate to achieve sufficient measurement accuracy. Reflections 3 dB below the direct sound level can cause sizeable peaks and dips in the measured response due to phase interference with the direct sound.

For the peaks and dips caused by reflections to be less than 1 dB, the reflected sound level should be more than 9 dB below the direct sound level at the output of the TEF's filter.

For example, suppose you're measuring the rear (180 degree) frequency response of a cardioid microphone. If the rear sensitivity of the cardioid microphone is -20



dB relative to the on-axis sensitivity, then the reflected sound level should be at least 9 dB less, or -29 dB relative to the direct sound level at the microphone, for +/- 1 dB accuracy.

The farther a reflective surface is from the center of the space window, the more its reflections are attenuated by the tracking filter. The table below shows reflection attenuation versus the distance of the reflective surface from the center of the space window:

Distance of surface to center space window, in number of space windows.	Attenuation of reflection, dB
1	3.2
2	13.7
3	33.7
4	72.5

Inverse-square law attenuation further reduces the level of reflections.

The table shows that, for maximum accuracy, reflective surfaces should be well outside the space-window ellipsoid.

The relation between resolution frequency and space window is:

$$F = C/D$$

where F is the resolution frequency in hertz

C is the speed of sound in feet per-second

D is the space window in feet.

If we want to measure down to 100 Hz, we need a space window of roughly 10 feet, or a clear space 5 feet around the microphone and loudspeaker (from the formula $F = C/D$). Check that the path length of each room reflection exceeds the direct-sound path by more than one wavelength of the lowest frequency to be measured.

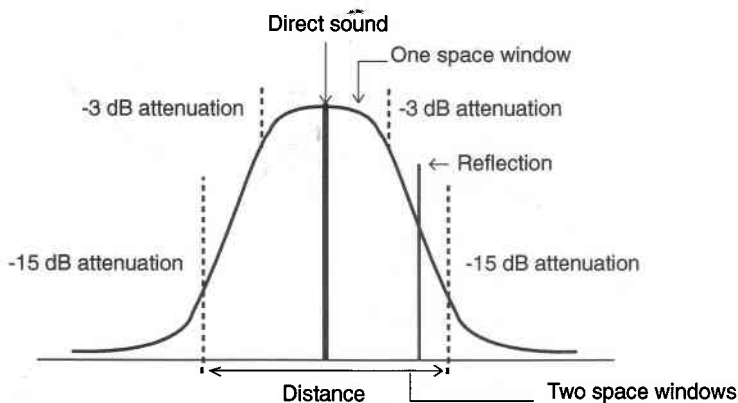


Illustration B-5
The space window

Appendix C: Data interpretation

Is it reasonable?

It has been said that in order to properly measure a device, you must know all there is to know about it. But, if you already know all there is to know about it, then why do you need to measure it? Obviously, when we measure something, we *don't* know all there is to know about it. That's why we are measuring it.

Because we don't know it all, we must be careful when we measure. We must convince ourselves that the results of our measurements are real and correct. When interpreting the data presented by the TEF Analyzer (or any measurement system for that matter), you must always ask yourself: "Is this reasonable?" If you are measuring the low frequency response of a 4-inch loudspeaker, and a huge bump appears at 60 Hz, is this reasonable? Most likely it is not.

When you see something like this, your first thought should be: "What did I do wrong?" In the above example, you would probably start looking for a source of 60 Hz hum in your measurement setup.

If you have checked and double checked and are convinced that your measurement is correct, it probably is. However, it should take a lot of convincing to believe that a 4-inch loudspeaker is putting out large amounts of 60 Hz.

Is it repeatable?

One test that can and should be made to determine the legitimacy of a measurement, is to check its repeatability. You should be able to recreate and repeat a valid measurement of a real phenomenon with the same results each time. The TEF Analyzer provides a very handy tool for checking the repeatability of a measurement—the Difference mode.

To check a measurement that you have just made, turn on the Difference mode and repeat the measurement. If it is repeatable, the difference should show virtually no variation between the two measurements.

Illustration C-1 is a difference with poor repeatability. There are variations of over 6 dB across the entire screen.

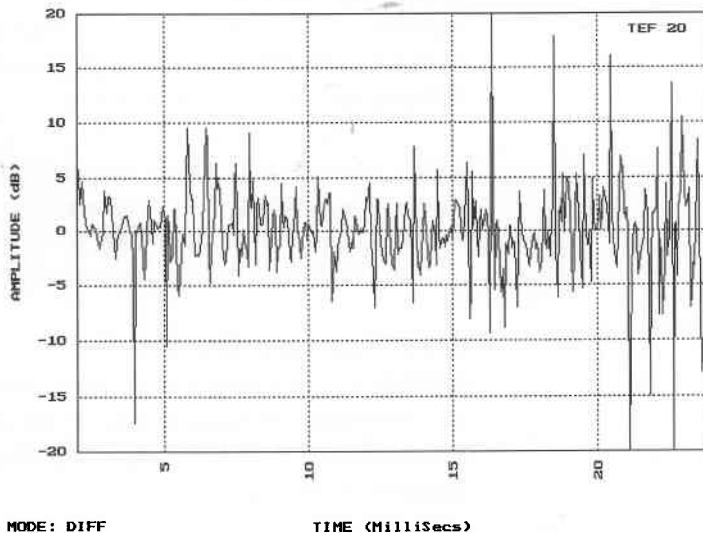


Illustration C-1 A difference with poor repeatability



Illustration C-2 is a difference showing good repeatability. There is very little difference anywhere in the display.

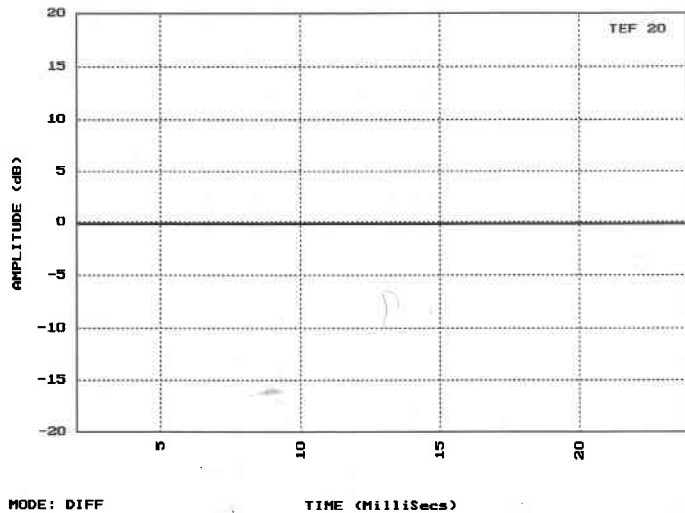


Illustration C-2 A difference with good repeatability

Another way to check for repeatability is to turn on overlay mode and make several measurements. The data from each of the measurements should lie right on top of each other.

Some things to watch for

There are some common errors which can easily creep into our measurements. If we can learn to recognize their telltale signs, we will be able to immediately spot them and take action to eliminate them.

Whenever two similar signals arrive at our measuring microphone with nearly the same level, but at slightly different times, they will interfere with each other and cause anomalies in the measured response. This will be the situation when we have set too large a time window and a reflection is measured along with the signal we are really interested in.

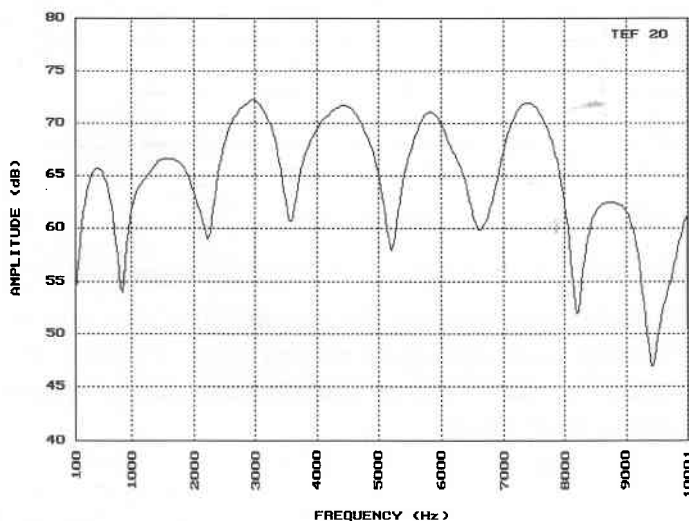


Illustration C-3 shows the telltale signature of two signals being measured at once. Notice that the peaks and notches are evenly spaced. Each notch is approximately 1389 Hz apart as is each peak. Widely spaced notches indicate that the signals are arriving fairly close

*Illustration C-3
Two signals being
measured at once*

together. Notches close together indicate that the signals are separated farther in time.

Illustration C-3 has a linear scale. On a logarithmic scale, this signature of two interfering signals, would be very difficult to see. This is one of the reasons for the growing popularity of linear scales.

This type of interference pattern may be caused by the direct sound from a loudspeaker and a reflection, or it could be the result of two drivers covering the same frequency range but emitting their sounds at slightly different times.

When making frequency response measurements, you must take the frequency resolution into account. Remember that in order to reliably measure a frequency, you must have enough time to observe at least one full period. If we have a frequency resolution of 500 Hz, the resulting time resolution is 2 msec. The period of a 500 Hz sound wave is 2 msec. This means that with 500 Hz of frequency resolution, all frequencies below 500 Hz are not being reliably measured.

The analyzer will display what may look like reasonable data below 500 Hz and, in fact, we can consider a part of it to be reliable. A rule of thumb is that from a frequency that is equal to $1/2$ of the frequency resolution on up, the data is reliable. With 500 Hz of frequency resolution, we can consider everything from 250 Hz on up to be reliable.

Noise is another problem to watch out for. TDS is highly immune to noise and, unless you are using high sweep rates in a very noisy environment, noise should not be a problem in TEF measurements. However, the ETC measurement is more susceptible to noise due to the wider filter bandwidths that are used.

Illustration C-4 shows a noisy ETC. Although the direct sound is easily seen at a level of 42 dB, the rest of the display is noise. If all you were interested in was the time of arrival and the level of the direct sound, it's there. However, if you were trying to measure the room, you would have to increase the level from the loudspeaker in order to bring the room's reflections up out of the noise.

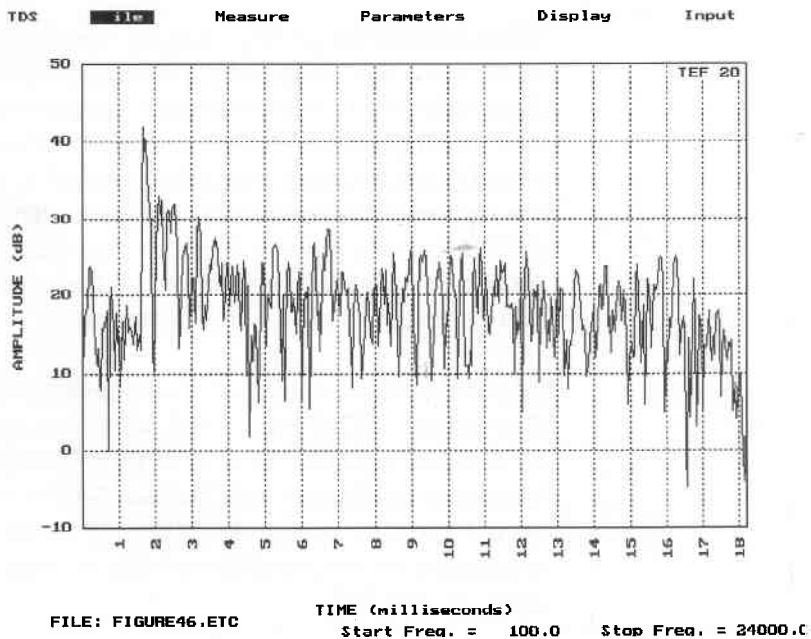


Illustration C-4
A noisy ETC showing the direct sound at 42 dB.

Appendix D: On-screen messages

This appendix lists some on-screen messages that you may encounter while using *SoundLab*. Other messages may also appear with possible solutions. Follow the on-screen instructions that accompany these messages to resolve problems. Messages listed in this section appear in alphabetical order, according to the first word in each message.

Awaiting TEF trigger

TEF is waiting for trigger signal at the external trigger connector.

Bandwidth should be $\geq \sqrt{\text{sweep rate}}$

A bandwidth that is greater than the square root of the sweep rate is not a TDS measurement and therefore does not reject reflections.

Can't difference...Number of points doesn't match

Both files must contain the same number of data points in order to difference them.

Can't difference...Different file type

You can't difference a time and a frequency file.

Can not save +1 this file type

You can't use *Save +1* with STI, RASTI nor 3D.

Check Communications or

Check Communication Port

This message appears when the host PC fails to communicate with the TEF analyzer. Check the match between the hardware settings (Section 5, *TEF 20 Analyzer manual*), and the selections made under the *Communication* sub-menu.

For serial communications, check the baud and COM port settings. For HI, check the base address setting. Also, either interface could have a faulty cable.

Turn ac power to the TEF *off* and then *on* or press *Reset* on the analyzer front panel after making changes.

Cursor is Off or Cursor is On

These messages are seen only when the data screen is empty and the cursor status has changed, or when test parameters have changed. Cursors are not visible when the screen data does not match the test parameters.

Delay before test ## seconds

The amount of time that remains before the test begins.

Disk error

Path was not found

Check the validity of the entries under drive and directory.

Drive was not found

Check the validity of the entries under drive and directory.

Disk is write protected

Remove write protection.

Drive is not ready

Be sure disk is in drive.

Write fault

Possibly a bad disk or a faulty disk drive.

Disk is full

Change disks or delete files.

Doing Noise Test 63 8000 Hz

This message appears during an NC test and the NC portion of STI and RASTI measurements.

Drive not ready or illegal file information

Check under *drive* and *directory* to see that *drive* and *path* are correct. Is there a disk in the drive?

Error file was not saved

Most likely there was no file name entered. Use *Save As*.

Error in data file

There is an error on a parameter value. This error should never occur, but, if it does exit the program and begin again.

Error writing data file

There was an error in writing to disk. The disk may be full or not available.

Insufficient disk space

Not enough space on disk. Change disks or delete some files.

Invalid path

Check the validity of the entries under *Drive and Directory*.

Loading will delete current data

Loading a configuration file will change the parameters thus invalidating the current data; therefore the data will be deleted.

Measure halted. Path is invalid

Check the validity of the entries under *Drive and Directory*.

Measurement interrupted with ### curves saved

When the 3D measurement is interrupted choosing the *Continue* button will continue the set of measure at the curve number that was interrupted. Choosing the *Stop* button will cause the measurement to cease and update the number of curves to match the number that have already been made. Choosing *Ease* will erase all of the curves that have already been saved for this measurement.

Min = 0 Max = 60.0

This message is seen only in the *Settings* sub-menu. Input gain can range from 0 to 60 in 4 dB increments.

Mode: Diff

Appears in the lower left of the screen when *Difference-On* has been selected in the *Display* menu.

Mode: Ovr

Appears in the lower left of the screen when *Overlay-On* has been selected in the *Display* menu.

No TCF files Found

Sound Lab didn't find a configuration file in the current path.

Obtain STI data before adding noise

You can't add noise to an STI measurement before one has been made or a file opened.

Overload Occurred

The input signal was too high. Reduce gain at the input, or lower the signal from the system under test.

Path/filename exists. Overwrite (Y/N)?

Appears anytime a file in the current path has the same name entered in the *Save As* menu. If you answer *Y*, the old data will be deleted and the new data will be saved.

Answer *N* and then change the name in the *Save As* menu.

No data current

Some instructions can not be executed without data existing in memory.

No data available

There was no data in memory to save.

Not enough memory for the phase cursor

There was not enough free RAM found to create the phase data array. Add more RAM to your computer or remove background programs such as memory managers or print queues.

Not enough memory for the file

Disk is too full. Use a different disk or delete some files.

Not enough memory for Nyquist

Not enough memory to create Nyquist data. Add more RAM to your computer or get rid of background tasks such as memory managers or print queues.

Not enough memory for cursor

Not enough memory to create cursor data. Add more RAM to your computer or get rid of background tasks such as memory managers or print queues.

Not enough memory for smoothing

Not enough memory to create smoothed data. Add more RAM to your computer or get rid of background tasks such as memory managers or print queues.

Not enough memory for test

Not enough memory to contain the data for this test. Add more RAM to your computer or get rid of background tasks such as memory managers or print queues.

Parameters may not be changed while overlay or difference mode is on

Turn off *Difference* and *Overlay* mode before changing parameters.

Performing FFT

This message appears during STI or RASTI measurements. An FFT is a complex operation which may take several minutes on computers without math co-processors. This message also appears when changing the noise with *Add Noise*.

Receiving Data

This message indicates the PC is receiving test data from the TEF over the HI or serial port.

Resolution must be greater than 0

A resolution of zero is a physical impossibility.

RT60 cursor is on or RT60 cursor is off

These messages are seen only when the data screen is empty and the cursor status has changed. Cursors are not visible when the screen data does not match the test parameters.

RT60 out of Range

This message is seen when *Sound Lab* places the RT60 cursors in a way that would result in an increasing slope. If this condition were true, the sound field would be increasing, not decaying. To correct this problem, manually place the cursors as described in the section titled *Making %ALcons and RT60 measurements*.

Start frequency should be \geq Bandwidth

A *Bandwidth* that is less than the bandwidth will cause an invalid response for the low frequency end of the measurement.

Test Interrupted

During STI calibration, this indicates the automatic sweep for calibrating level has been stopped by the user. Press *Enter* to proceed to the equalization test.

During ETC or TDS sweeps, this indicates *Escape* was pressed.

Testing

This message appears during a TEF sweep. It is usually followed by *Receiving Data*.

Test Timed Out

The PC did not receive the test data from the analyzer in the required amount of time. See the *Check Communications* section found on previous pages of this section.

The printer is not ready/check the printer and try again

Be sure the cable is connected to the printer, it is turned on, on line, and plugged in.

The TEF is not responding! Reset the TEF and try again

You have lost communications with the TEF. Make sure the correct cable is connected and the communication settings are correct for the cable used and the hardware switch settings.

There is not data available to print

Make a measurement or load data with *Open Old* before printing.

Time span should not exceed sweep time

Reduce time span or increase sweep time.

Appendix E: ASCII file formats

TEF ASCII file formats consist of two or more blocks of information. Usually, these consist of a header block and a data block. Information in both of these blocks is needed to process TEF data with spreadsheets and high level programs such as BASIC or Pascal.

Header block format

Each line of information within the header blocks is usually in the following format:

- A parameter label in double quotes
- An ASCII tab character (hexadecimal 09)
- A value. The value is in double quotes if it is alphanumeric
- An ASCII carriage-return character (hexadecimal 0D) for Macintosh files and a carriage-return and line-feed for MS-DOS files (hexadecimal 0D 0A).

Macintosh line format:

"Parameter label"<tab>numeric value<cr>

MS-DOS line format:

"Parameter label"<tab>numeric value<cr><lf>

Data block format

The data block is usually stored as a voltage measured by the TEF. Most files store the voltage as complex number pairs. See the data section for each file type for the exact storage format.

ETC file format

The ETC header

The first part of the ETC file is the header block. The header is a record of all analyzer settings that went into making the test. Also included in the header are operator comments, test location, and the date the test was made.

The ETC header has 29 lines. Line 29 is always "Data =".

The ETC data

The second part of the ETC file is the collected data. The data is stored as the voltage measured by the TEF independent of the preamp gain. The number of lines in the data part of the file always equals the number of samples selected when the test was made (512, 1024, 2048, 4096, or 8192 lines). Each line consists of the following items:

- A real number in scientific notation
- A tab character (hexadecimal 09)
- An imaginary number in scientific notation
- An ASCII carriage-return character (hexadecimal 0D) for Macintosh files and, a carriage-return and line-feed for MS-DOS files (hexadecimal 0D 0A).

Notes on ETC files

Parameter label	Value	Comments
"ETC"		Always use this key word for ETC measurements.
"Title = "	"	Maximum of 50 characters.
"Name = "	"	Maximum of 50 characters.
"Date = "	"	Maximum of 50 characters.
"Location = "	"	Maximum of 50 characters.
"Description = "	"	Maximum of 128 characters.
"Save Rate (k) = "	47	
"Number of Samples = "	1024	
"Start Frequency = "	1.78E+03	
"Stop Frequency = "	2.22E+03	
"Sweep Time = "	4.01E+03	
"Sweep Rate = "	9.57E+03	
"Bandwidth = "	0.00E+00	
"Receive Delay = "	0.00E+00	
"ETC Data Window Type = "	"HAMMING"	ETC supports only five types of data windows. All must be entered in upper case letters. "BLACKMAN" "HAMMING" "HANNING" "KAISER" "RECTANGULAR"
"ETC Data Window Beta = "	2.00E+00	Beta is used only for Kaiser windows and must be a value between 2 and 15. Beta is always present regardless of the window type.
"Reference Unit = "	"Pascal"	
"Volts Per Reference Unit = "	2.26E-03	
"Zero dB Reference Value = "	2.00E-05	
"Propagation Speed = "	1.13E+03	

Parameter label	Value	Comments
"Distance Unit = "	"FEET"	Distance Units can only be expressed in one of four key words. Note all are upper case. "FEET" "INCH" "METER" "CENTIMETER"
"Channel = "	3	0 = Line input B 1 = Microphone input B 2 = Line input A 3 = Microphone input A
"Preamp Gain A = "	60	This is the output level used for this ETC. The output level value cannot be positive, and is expressed in dB below 1.0 volt rms. This value is valid only when the front panel level control is in the <i>Cal</i> position.
"Preamp Gain B = "	0	
"Output Level = "	-4.0007	
"Data format = "	"ASCII"	Data types can only be expressed in two key words, "SINGLE" and "DDOUBLE". There is currently no support for "DDOUBLE".
"Data type = "	"SINGLE"	
"Data count = "	1023	This is a zero based count. "Data count" is always equal to "Number of samples" -1.
"Data = "		This is always the last line of the header and precedes the data.

ALTARFB-ETC.TXT			
	A	B	
1	ETC		
2	Title =	ETC Taken To Show F	
3	Name =	LJS/JAB	
4	Date =	3/7/91 15:49	
5	Location =	Coalbush Church: So	
6	Description =	1 Second ETC, Center	
7	Save Rate (k) =	47	
8	Number of Samples =	1024	
9	Start Frequency =	1.78E+03	
10	Stop Frequency =	2.22E+03	
11	Sweep Time =	4.01E+00	
12	Sweep Rate =	9.57E+03	
13	Bandwidth =	0.00E+00	
14	Receive Delay =	0.00E+00	
15	ETC Data Window Type =	HAMMING	
16	ETC Data Window Beta =	2.00E+00	
17	Reference Unit =	Pascal	
18	Volts Per Reference Unit =	2.26E-03	
19	Zero dB Reference Value =	2.00E-05	
20	Propagation Speed =	1.13E+03	
21	Distance Unit =	feet	
22	Channel =	3	
23	Preamplifier Gain A =	60	
24	Preamplifier Gain B =	0	
25	Output Level =	0	
26	Data format =	ASCII	
27	Data type =	SINGLE	
28	Data count =	1023	
29	Data =		
30		3.83E-06	2.57E-06
31		1.77E-06	1.29E-06
1052		3.15E-06	4.59E-06
1053		3.15E-06	4.59E-06
1054			

Illustration E-1 ETC ASCII file displayed with a spreadsheet. The spreadsheet does not display the double quotes in the header.



NC ASCII files

The NC header

The first part of the NC file is the header information. The header is a record of all analyzer settings that contributed to the test. Also included in the header are operator comments, test location, and the date the test was made. See *Header information format* at the beginning of this section.

Line 16, the last line of the header, is always "Data =".

The NC data

The second block of the NC file is eight lines of collected data. The data is stored as sound pressure level, expressed in dB. These values include factors entered in the *Calibration* sub-menu of Sound Lab.

Each line of NC data contains the following items:

- A sound pressure level in scientific notation
- A tab character (hexadecimal 09)
- Zero, expressed in scientific notation
- An ASCII carriage-return character (hexadecimal 0D) for Macintosh files and a carriage-return and line-feed for MS-DOS files (hexadecimal 0D 0A).

Notes on NC files

Parameter label	Value	Comments
"NC"		Always use "NC" as the test type key word for NC measurements.
"Title = "	64??	Maximum of 50 characters.
"Name = "	64??	Maximum of 50 characters.
"Date = "	64??	Maximum of 50 characters.
"Location = "	64??	Maximum of 50 characters.
"Description = "	" "	Maximum of 128 characters.
"Integration time = "	4	This value must be between 1 and 9,999. Default setting is 4.
"Number of Samples = "	8	Number of Samples is fixed at 8.
"Volts Per Reference Unit = "	2.26E-03	
"Channel = "	3	0 = Line input B 1 = Microphone input B 2 = Line input A 3 = Microphone input A
"Preamp Gain A = "	60	
"Preamp Gain B = "	0	
"Data format = "	"ASCII"	
"Data type = "	"SINGLE"	
"Data count = "	7	Data count is fixed at 7.
"Data = "		Data is in dB SPL.
4.16E+01	0.000000E+0	63 Hz band
4.00E+01	0.000000E+0	125 Hz band
3.15E+01	0.000000E+0	250 Hz band
2.70E+01	0.000000E+0	500 Hz band
2.22E+01	0.000000E+0	1000 Hz band
2.09E+01	0.000000E+0	2000 Hz band
1.33E+01	0.000000E+0	4000 Hz band
3.35E+01	0.000000E+0	8000 Hz band

CFANOFF-NC.TXT				
	A	B	C	D
1	NC			
2	Title =	Noise Criteria Measurement With Ceiling Fans Off		
3	Name =	LJS/JAB		
4	Date =	3/7/91 14:52		
5	Location =	Coalbush Church: South Bend, IN		
6	Description =	Mic Just In Front Of Balcony On Main Floor		
7	Integration Time =	4		
8	Number of Samples =	8		
9	Volts Per Reference Unit =	2.26E-03		
10	Channel =	3		
11	Preamplifier Gain A =	60		
12	Preamplifier Gain B =	0		
13	Data format =	ASCII		
14	Data type =	SINGLE		
15	Data count =	7		
16	Data =			
17		4.16E+01	0.00E+00	
18		4.00E+01	0.00E+00	
19		3.15E+01	0.00E+00	
20		2.70E+01	0.00E+00	
21		2.22E+01	0.00E+00	
22		2.09E+01	0.00E+00	
23		1.33E+01	0.00E+00	
24		3.35E-07	0.00E+00	
25				

Illustration E-2 NC ASCII file displayed in a spreadsheet. The double quotes are not displayed by the spreadsheet.



TDS ASCII files

The TDS header

The first part of the TDS file is the header information. The header is a record of all analyzer settings that went into making the test. Also included in the header are operator comments, test location, and the date the test was made. Line 31, the last line of the header, is always "Data =".

The TDS data

The second part of the TDS file is the collected data. The data is stored as the voltage measured by the TEF independent of the preamp gain. The number of lines in the data part of the file always equals the number of samples selected when the test was made (512, 1024, 2048, 4096, or 8192 lines).

Each line consists of the following items:

- A real number in scientific notation
- A tab character (hexadecimal 09)
- An imaginary number in scientific notation
- An ASCII carriage-return character (hexadecimal 0D) for Macintosh files and a carriage-return and line-feed for MS-DOS files (hexadecimal 0D 0A).

ALTARFB-TDS.TXT			
	A	B	
2	Title =		
3	Name =	LJS/JAB	
4	Date =	3/7/91 18:01	
5	Location =	South Bend	
6	Description =	Church	
7	Save Rate (k) =	1	
8	Number of Samples =	1024	
9	Start Frequency =	2.00E+02	
10	Stop Frequency =	1.50E+04	
11	Sweep Time =	1.92E+00	
12	Sweep Rate =	7.71E+03	
13	Bandwidth =	4.63E+01	
14	Time Resolution =	3.33E+00	
15	Distance Resolution =	3.77E+00	
16	Frequency Resolution =	3.00E+02	
17	Critical Bandwidth On =	ON	
18	Receive Delay =	4.05E+01	
19	Reference Unit =	Pascal	
20	Volts Per Reference Unit =	2.26E-03	
21	Zero dB Reference Value =	2.00E-05	
22	Propagation Speed =	1.13E+03	
23	Distance Unit =	FEET	
24	Channel =	3	
25	Preamp Gain A =	40	
26	Preamp Gain B =	0	
27	Output Level =	0	
28	Data format =	ASCII	
29	Data type =	SINGLE	
30	Data count =	1023	
31	Data =		

*Illustration E-9 TDS ASCII header displayed in a spreadsheet.
The double quotes are not displayed by the spreadsheet*

Notes on TDS files

Parameter label	Value	Comments
"TDS"		Always use this key word for TDS measurements.
"Title = "	44.33	Maximum of 50 characters.
"Name = "	44.33	Maximum of 50 characters.
"Date = "	""	Maximum of 50 characters.
"Location = "	""	Maximum of 50 characters.
"Description = "	""	Maximum of 128 characters.
"Save Rate (k) = "	19	
"Number of Samples = "	1024	
"Start Frequency = "	2.00E-02	
"Stop Frequency = "	1.50E+04	
"Sweep Time = "	1.92E+00	
"Sweep Rate = "	7.71E+03	
"Bandwidth = "	4.63E+01	
"Time Resolution = "	3.33E+00	
"Distance Resolution = "	3.77E+00	
"Frequency Resolution = "	3.00E+02	
"Critical Bandwidth On = "	"ON"	If Critical Bandwidth is turned on, then use the key word "ON", otherwise use the key word "OFF".
"Receive Delay = "	4.05E-01	
"Reference Unit = "	"Pascal"	
"Volts Per Reference Unit = "	2.26E-03	
"Zero dB Reference Value = "	2.00E-05	
"Propagation Speed = "	1.13E+03	
"Distance Unit = "	"FEET"	Distance units are key words and can only be expressed in four words. Note all are in upper case letters. "FEET" "INCH" "METER" "CENTIMETER"

Parameter label	Value	Comments
"Channel = "	3	0 = Line input B 1 = Microphone input B 2 = Line input A 3 = Microphone input A
"Preamp Gain A = "	40	
"Preamp Gain B = "	0	
"Output Level = "	0	This is the output level used for this TDS. The output level value cannot be positive, and is expressed in dB below 1.0 volt rms. This value is valid only when the front panel level control is in the <i>Cal</i> position.
"Data format = "	"ASCII"	
"Data type = "	"SINGLE"	Data types are key words and can only be expressed in two key words. There is currently no support for "DDOUBLE"
		"SINGLE"
		"DDOUBLE"
"Data count = "	1023	This is a zero based count. "Data count" is always equal to "Number of samples" -1.
"Data = "		
-2.678063e-6	7.765387e-7	
-2.976732e-6	-3.902605e-6	
-5.953464e-6	-2.588463e-6	

<i>Analyzer settings</i>
<i>Summary table</i>
<i>Noise data</i>
<i>ETC1 header</i>
<i>ETC1 data</i>
• • •
<i>ETC7 header</i>
<i>ETC7 data</i>

*Illustration E-10
The blocks of the
STI ASCII file
format.*

STI and RASTI file formats

STI and RASTI files are composed of several blocks of data. Some of the blocks contain header information and some contain data.

STI analyzer settings block

Both RASTI and STI measurements store a record of all analyzer hardware settings in the analyzer settings block. Also included in the settings block are operator comments, test location, and the date the test was made. These settings are common between the NC and ETC measurements. The settings block has 14 lines.

STI summary table block

The summary table block contains the column headings for the STI display as well as the values for the display. An ASCII tab character (hexadecimal 09) separates the columns.

If the measurement is an "STI" or "NOISELESS_STI", the display block will contain data for all seven frequency bands. If the measurement was a "RASTI" or "NOISELESS RASTI" measurement, only two bands of data will appear in the summary table block.

STI ETC header block

The ETC header in STI and RASTI files contain only those analyzer settings that would appear in an ETC *Parameters* sub-menu. The parameters used for the STI and RASTI tests are fixed by the program and cannot be changed by the operator.

The last line of the ETC header block is always "Data =".

STI ETC data blocks

The second part of the ETC file is the collected data. The number of samples is different for each frequency band of the STI measurement. The data is stored as complex pairs, identical to the storage format for ETC files.

Notes on STI analyzer settings block

Parameter label	Value	Comments
"STI"		Always use this key word for STI and RASTI measurements.
"FULL_STI"		Must be one of the following four key words: "FULL_STI" "NOISELESS_STI" "FULL_RASTI" "NOISELESS_RASTI"
"NC_NOISE"		Can be one of the following three key words: "NC_NOISE"—If the measurement was a "FULL_STI" or "FULL_RASTI", and the noise source was from an actual STI or RASTI measurement or the noise was brought in from a NC measurement, then use the "NC_NOISE" key word. "TABLED_NOISE"—If the measurement was a "FULL_STI" or "FULL_RASTI" with noise, and the noise source was from user-entered table list, then use the "TABLED_NOISE" key word. "NO_NOISE"—If the measurement was a "NOISELESS_STI" or "NOISELESS_RASTI" (a measurement without noise, then use the "NO_NOISE" key word).

Parameter label	Value	Comments
"Title = "	" "	
"Name = "	" "	
"Date = "	" "	
"Location = "	" "	
"Description = "	" "	
"Volts Per Reference Unit = "	.01	
"Propagation Speed = "	1130.00	
"Distance Unit = "	"FEET"	Distance Units are key words and can only be expressed in one of four key words. Note all are upper case. "FEET" "INCH" "METER" "CENTIMETER"
"Channel = "	3	0 = Line input B 1 = Microphone input B 2 = Line input A 3 = Microphone input A
"Preamp Gain A = "	48	
"Preamp Gain B = "	0	

ALTARFB-STI.TXT			
	A	B	C
1	STI		
2	FULL_STI		
3	NC_NOISE		
4	Title =	STI Taken Through Altar Mic	
5	Name =	LJS/JAB	
6	Date =	3/7/91 15:49	
7	Location =	Coalbush Church: South Bend, IN	
8	Description =	Main System Used To Excite The Room	
9	Volts Per Reference Unit =	0.01	
10	Propagation Speed =	1130	
11	Distance Unit =	FEET	
12	Channel =	3	
13	Preamp Gain A =	40	
14	Preamp Gain B =	0	

Illustration E-11 STI and RASTI analyzer settings block



Notes on STI summary table block

"FREQUENCY"	"STI"	"EARLY RT60"	"S/N RATIO"
125 Hz	0.85	0.3	10.5
250 Hz	0.7	0.61	6
500 Hz	0.53	1.13	.09
1000 Hz	0.47	2.07	-1
2000 Hz	0.49	1.71	-.04
4000 Hz	0.35	3.37	-4.4
8000 Hz	0.34	5.34	-4.8
"OVERALL STI = "	0.46		
"EQUIVALENT S/N RATIO ="	1		
"EQUIVALENT EARLY RT60 ="	2.1		
"SUBJECTIVE EVALUATION = "	"FAIR"		

Subjective evaluations are key words and can only be expressed in five key words.

"EXCELLENT"

"GOOD"

"FAIR"

"POOR"

"BAD"

ALTARFB-STI.TXT				
	A	B	C	D
15	FREQUENCY	STI	EARLY RT60	S/N RATIO
16	125 Hz	0.85	0.3	10.5
17	250 Hz	0.7	0.61	6
18	500 Hz	0.53	1.31	0.9
19	1000 Hz	0.47	2.07	-1
20	2000 Hz	0.49	1.71	-0.4
21	4000 Hz	0.35	3.37	-4.4
22	8000 Hz	0.34	5.34	-4.8
23	OVERALL STI =	0.46		
24	EQUIVALENT S/N RATIO =	1		
25	EQUIVALENT EARLY RT60 =	2.1		
26	SUBJECTIVE EVALUATION =	FAIR		

Illustration E-12 STI and RASTI summary table block

Notes on STI noise data block

Parameter label	Value	Comments
"NOISE DATA = "		<p>This is the key word for the start of the noise data summary. The eight values are always shown regardless of whether a "NOISELESS_STI" or "NOISELESS_RASTI" measurement was performed. If a "NOISELESS_STI" or "NOISELESS_RASTI" measurement was performed, the values are expected to be all zero (0.000000e+0).</p> <p>If the measurement is a "FULL_RASTI" and the noise was not input from a table or from a NC file, then element four and element six will be the only elements with values as shown below.</p>
1.70E+01		
2.90E+01		
3.50E+01		
3.40E+01		Used only for RASTI
3.00E+01		
2.80E+01		Used only for RASTI
2.40E+01		
2.00E+01		
"Integration time = "	4	<p>The STI setting is always 4 for integration time. If noise data was imported from an NC file, this value will always fall between 1 and 9,999.</p>

ALTARFB-STI.TXT			
	A	B	
37	ETC1		
38	Save Rate (k) =	94	
39	Number of Samples =	512	
40	Start Frequency =	11	
41	Stop Frequency =	228	
42	Sweep Time =	4.01	
43	Sweep Rate =	54.1	
44	Bandwidth =	0	
45	Receive Delay =	0	
46	Output Level =	-2.5	
47	Data format =	ASCII	
48	Data type =	SINGLE	
49	Data count =	511	
50	Data =		
51	1.42E-06	-1.82E-06	
52	1.58E-06	1.40E-06	
561	2.31E-06	2.34E-06	
562	2.31E-06	2.34E-06	
563	ETC2		

Illustration E-14 STI and RASTI ETC header block.

Notes on STI ETC block

Parameter label	Value	Comments
"ETC1"		This is the key word for the start of the ETC data header. If the measurement is a "FULL_STI" or "NOISELESS_STI" there will be six sets of headers and ETC data. Each new ETC set will begin with the "ETCx" key word. (x) will be a number from one to seven as shown below . "ETC1" "ETC2" "ETC3" "ETC4" "ETC5" "ETC6" "ETC7"
		If the measurement is a "FULL_RASTI" or "NOISELESS_RASTI" there will be two sets of headers and ETC data. The first will be "ETC3" and the second "ETC5".
"ETC3"		
"ETC5"		
"Save Rate (k) = "	94	
"Number of Samples = "	512	
"Start Frequency = "	11.0	
"Stop Frequency = "	228.0	
"Sweep Time = "	4.01	
"Sweep Rate = "	54.1	
"Bandwidth = "	0.0	
"Receive Delay = "	0.0000	

Parameter label	Value	Comments
"Output Level = "	-2.5	This is a key word of the actual output level used for this ETC. This value contains <i>both</i> the BandGainLowerFactors coded into the STI application to simulate human voice <i>and</i> SpeakerEQOctaveSPL values collected from the talker equalization for the appropriate band. The values cannot be positive, and are in dB down from 1.0 vrms.
"Data format = "	"ASCII"	Data formats are key words and can only be expressed in two key words.
"Data type = "	"SINGLE"	
"ASCII"		Data types are key words and can only be expressed in key words. There is currently no support for "DDOUBLE".
"SINGLE"		
"Data count = "	511	This is a zero based count. "Data count" is always equal to "Number of samples" -1.
"Data = "		
-2.643692e-5	-2.095901e-5	
2.891838e-5	3.915130e-5	
-1.943973e-5	-3.417925e-5	

Parameter label	Value	Comments
"ETC2"		
"Save Rate (k) = "	47	
"Number of Samples = "	512	
"Start Frequency = "	141.0	
"Stop Frequency = "	358.0	
"Sweep Time = "	2.01	
"Sweep Rate = "	1496.0	
"Bandwidth = "	0.0	
"Receive Delay = "	0.0000	
"Output Level = "	-9.1014	
"Data format = "	"ASCII"	
"Data type = "	"SINGLE"	
"Data count = "	511	
"Data = "		
9.944027e-6	2.611652e-5	
-1.224643e-5	-2.667577e-5	
7.479048e-6	1.317536e-5	
"ETC3"		
"ETC4"		
"ETC5"		
"ETC6"		
"ETC7"		

3D file format

The 3D master file header

The 3D master file (with extension .3D) contains information that is common to all individual tests in the 3D set, such as the *test title*, *date* and *location*, *description* of the test and *number of measurements* or TDS files in the measurement. The lines for *Third Axis* contain labeling and value information for *Start* time, *Stop* time and *Time* step. The *Max Y Data* value is the largest data value in all TDS files. The *Min Y Data* is the minimum data value in all TDS files.

The associated data for each of the measurements in the set accompanies the file with the same name plus the extensions .001, .002, .003...etc.

See *TDS ASCII files, E-10* for information on individual TDS files.

TEST.3D		
	A	B
1	3D20A	
2	TDS	
3	Title =	Altar Mic
4	Name =	LJS/JAB
5	Date =	3/7/91 18:01
6	Location =	Coal Church
7	Description =	PCC 160
8	Number of Measurements =	36
9	Third Axis Label =	TIME
10	Third Axis Start =	0.00E+00
11	Third Axis Stop =	1.62E+01
12	Third Axis Step =	4.50E-01
13	Max Y Data =	7.00E+01
14	Min Y Data =	-1.00E+01

Illustration E-15 3D master file in a spreadsheet.



Notes on 3D files

Parameter label	Value	Comments
"3D20A"		File version number.
"TDS"		Always use this key word for 3D measurements.
"Title ="	" "	Maximum of 50 characters.
"Name ="	" "	Maximum of 50 characters.
"Date ="	" "	Maximum of 50 characters.
"Location ="	" "	Maximum of 50 characters.
"Description ="	" "	Maximum of 128 characters.
"Number of Measurements ="	36	From 2-100 measurements.
"Third Axis Label ="	TIME	
"Third Axis Start ="	0.00E+00	The Start time value.
"Third Axis Stop ="	1.62E+01	The Stop time value
"Third Axis Step ="	4.50E-01	The Time step value.
"Max Y Data ="	7.00E+01	Largest data value in all curves.
"Min Y Data ="	-1.00E+01	Smallest data value in all curves.

Appendix F: Glossary

acoustic center—The point in space of the origin of sound; for a sound emitting transducer, the point from which the spherical waves appear to diverge as observed at remote points.

acoustic origin—the point in time at which the signal originates.

% AL_{CONS} —The measured percentage of Articulation Loss of Consonants by a listener. In TEF, Articulation scores are measured as percent of articulation loss of consonants in speech. % AL_{CONS} of 0 indicates perfect clarity and intelligibility with no loss of consonant understanding, while 10% and beyond is growing toward bad intelligibility, and 15% typically is the maximum loss acceptable.

A-weighting *see* Weighted.

ambience—Room acoustics, early reflections and reverberation. The audible sense of a room or environment surrounding a sound source.

ambient noise—Background noise associated with a given environment.

amplitude—In TEF measurements, the total summation of all sound energy over the total time of the measurement at all frequencies within the bandpass of the instrument. The sound energy at a given frequency over the entire time of the measurement. Amplitude can be measured as the sound pressure at a given instant of time at a given frequency. Amplitude is the maximum value of a field quantity in space or time.

analyzer—A device that divides a spectrum into a finite number of frequency bands and determines the relative magnitude of the energy in each band. TEF analyzers combine the capabilities of a computer, sweep oscillator, an accurate quartz timing clock, and a sweepable bandpass filter system to make TDS measurements. This hardware, coupled with controlling software, gives control of frequency, energy, and signal delay, along with the precision to measure and analyze the results. In addition, TEF analyzers allow extensive post-processing capabilities and storage of test data.

TEF analyzers can sweep linearly in time through a specified range of frequencies. The characteristics of this oscillator are its sweep rate in hertz per second, and its starting and stopping frequencies. The analyzer in the TEF systems linearly sweeps its tuning through a range of frequencies. Its characteristics are its sweep rate, bandwidth, and start frequency and stop frequency. Since this bandwidth is sweeping in time, it can also be described as a time window in seconds. The analyzer time aperture equals the bandwidth in frequency (hertz) divided by the sweep rate in Hertz per second. This window is proportional to bandwidth and is inversely proportional to sweep rate. In mathematical terms: $R_t = B/S$

Time Resolution = bandwidth/sweep rate

anechoic—Literally, without echo. A characteristic describing an environment whose boundaries effectively absorb all sound over the frequency range of interest, thereby creating a free field condition. By filtering out delayed reflections, the TEF can make anechoic (echo-free) measurements in a non-anechoic room.

articulation loss of consonants—A measure of speech intelligibility. The percentage of consonants heard incorrectly, strongly influenced by noise or excessive reverberation.

see: $\% AL_{\text{CONS}}$

attenuate, attenuation—The lessening of the sound signal level due to divergence, absorption, reflection, refraction, diffraction, etc. expressed in decibels. The decrease in sound level with distance in the direction of propagation. The reduction of the level of a speaker.

band-pass filter—A filter that passes a specified frequency band while all frequencies above and below this band are attenuated. *see:* bandwidth, center frequency.

bandwidth—The difference between the values of the frequencies where the filter's response has fallen by 3 dB. In TEF, the bandwidth of the tracking filter can be preset. The wider the bandwidth, the greater the "time window." *see:* time window, band-pass filter, center frequency.

comb filter, comb filter effect—A sequence of evenly spaced peaks or dips in the frequency response plot when viewed on linear scale caused by two or more identical signals which combine at near equal amplitudes but at slightly different time intervals.

complex wave—A wave with more than one frequency component.

compression—The portion of a sound wave in which molecules are pushed together, forming a region with higher-than-normal atmospheric pressure. Also, in signal processing, the reduction in dynamic range caused by a compressor.

coverage—the distribution of direct sound levels in a listening area.

coverage angle—The angle included between 6 dB down points of a sound source.

crest factor—The ratio of peak to rms values of a waveform.

critical distance—The distance from a sound source at which direct sound and reverberant sound are at the same level.

critical frequency—The frequency below which standing waves cause significant room modes.

dB—Abbreviation for decibel. *see*: decibel.

deadness—The lack of sound reflections in a room. The subjective judgment of how a room “sounds”, dependent upon the initial time delay gap and the ratio of direct sound level to the early reflection level.

decay rate, decay time—The rate at which the reverberant sound field decays in a room, measured in dB/second. Decay rate is related to reverberation time by $R_d = 60/RT_{60}$.

decibel (dB)—A power ratio. The unit of measurement of audio level. Ten times the logarithm of the ratio of two power levels. Twenty times the logarithm of the ratio of two voltages, currents or sound pressures.

dBV is decibels relative to 1 volt.

dBm is decibels relative to 1 milliwatt.

dBA is decibels, A weighted (see Weighted)

A decibel is commonly thought to be the smallest change in sound pressure level that the trained human ear can detect.

delay—The time interval between two signals.

Delay can be:

1. the signal delay time through a given component
2. the signal delay time from a loudspeaker to listener
3. the signal delay between two transducers (either microphones or loudspeakers.)
4. Any other signal delay in a sound system that exceeds that normally expected from a minimum delay system.



diffraction—The bending of a wave front around an obstacle in the sound field. *see:* reflection.

diffuse field—Sound field in which the sound pressure level is the same everywhere and the flow of energy is equally probable in all directions.

diffuser—A device to enhance the spreading of sound for even distribution of sound in an environment.

diffusion—The spreading of sound reflections to achieve an even distribution of sound in an environment.

direct sound—Sound that has traveled from the sound source to the observer and has encountered no reflecting surfaces. *see:* Q

directivity factor (Q)—The ratio of the sound pressure squared, radiated directly ahead of a sound source, to that sound pressure squared radiated in all directions.

discrete sound arrivals—Sound arrivals at the microphone or listening position that are separated in time.

domain—The X axis or independent variable in a measurement. *see:* time domain, frequency domain.

doppler effect—The change in the observed frequency of a wave caused by a change in the velocity of the sound source. An example of the doppler effect is the difference perceived in pitch of a car horn as it approaches. As it approaches, it appears to rise; as it passes and moves away, it appears to drop.

doubling—A special effect in which a signal is combined with its 15-to-35 millisecond delayed replica. This process mimics the sound of two identical voices or instruments playing in unison.

early decay time—The time for a sound to decay 10 dB from its original level. Short decay times cause music and speech to sound dry or muffled. Long decay times make speech unintelligible and difficult to understand. It is the figure that most closely approximates how the decay time “sounds” to the ear.

early sound, early reflections—Sound arriving within about 70 milliseconds of the direct sound.

echo—A sound wave which has been reflected or otherwise returned with sufficient magnitude and delay to be perceived as distinct from that directly transmitted. Echoes are perceived as distinct repetitions of the original sound. A sound delayed 90 milliseconds or more, combined with the original sound is sometimes considered an echo.

EFC, Energy Frequency Curve—A Frequency Response. A “snapshot” of all the energy returned in the frequency range of interest for a given amount of time. Frequency is displayed on the horizontal axis; magnitude on the vertical axis.

ellipsoid—a three dimensional ellipse. In TDS, the football-shaped space around the loudspeaker and microphone corresponding to points at which the TEF test tone is attenuated by 3 dB upon returning to the microphone. *see: space window*

ETC (Time response)—energy time curve—In TEF measurements, a display of all the energy returned during the time span of interest. Time is displayed on the horizontal axis; energy on the vertical axis. An ETC shows that "at this time, this much sound energy has arrived." An ETC indicates how energy comes out of, or is released from a system or device after it is hit with a sudden application of input energy confined to a given frequency band. ETC measurements quickly reveal not only the amplitude and the time of arrival but also the density of the reverberant field, its approach to exponential growth and decay, and the initial time delay gap. An ETC contains no frequency information other than the knowledge of the range being swept.

far field—The distribution of sound energy at a very much greater distance from a source than the linear dimensions of the source and in which the sound waves can be considered to be plane waves.

FFT—Fast Fourier Transform. An algorithm for rapidly computing the Fourier Transform.

flutter echo —A series of specific reflective returns caused by large surfaces being parallel to each other.

focused reflections—Sound energy concentrated by a curved surface. Focused reflections are usually louder than the normal reverberant field at a given time after the excitation has ceased. They can be caused by domed ceilings, curved surfaces, etc.

Fourier transform—A mathematical description of the relationship between functions of time and corresponding functions of frequency. It is a map to convert data from one domain into another. For example, if we have a signal that is a function of time—an impulse response—for example, then the Fourier Transform will convert that time domain data into frequency data yielding a signal that is a function of frequency—a frequency response. The inverse Fourier Transform will do just the opposite. It will give the time domain data from the frequency domain. The Fourier transform is executed by the computer in the TEF when making Energy Time Curves.

free field—An environment in which there are no reflective surfaces within the frequency region of interest.

frequency—The number of complete cycles or vibrations per unit of time, usually per second. The frequency of a wave (measured in hertz (Hz)) is equal to the velocity divided by the wavelength. A low-frequency sound (say, 100 Hz) has a low pitch; a high-frequency sound (say, 10,000 Hz) has a high pitch. Frequency is a measure of oftenness. The units of frequency are reciprocal of the units of time.

frequency resolution—*see*: resolution, frequency



frequency response—Amplitude versus frequency plot. In TEF measurements, energy density versus the frequency for a selected time window. When stated as a device specification, frequency response is the range of frequencies that an audio device will reproduce at an equal level (within a tolerance, such as ± 3 dB).

frequency span—That region of frequencies, from the lowest to the highest, over which the TEF sweeps for analysis and display.

frequencies of interest—*see:* frequency span.

FTC, Frequency Time Curve—A graph of time and frequency with magnitude displayed in the form of dB contour lines.

full scale time—The time span shown on the right end of the screen of an ETC measurement. It is dependent on the frequency span of the sweep and number of samples.

fundamental—The lowest frequency in a complex periodic wave.

gain—an increase in power. The ratio, expressed in decibels, between output power and input power of a system.

see: decibel

harmonic—An overtone whose frequency is a whole-number multiple of the fundamental frequency.

hertz—The unit of frequency representing cycles per second.

heterodyning—Mixing two frequencies together in order to produce two other frequencies equal to the sum and difference of the first two. For example, heterodyning a 100 kHz and a 10 kHz signal will produce a 110 kHz (sum frequency) and a 90 kHz (difference frequency) signal. In TEF, it means changing the frequency of the incoming signal (signal being analyzed) so that it is at the I.F. filter's center frequency.

highpass filter—A filter that passes frequencies above a certain frequency and attenuates frequencies below that same frequency. A low-cut filter.

Hz—Abbreviation for hertz.

impulse response—Sound pressure versus time measurement showing how a device responds to an impulse. A potential versus time measurement showing how the potential of a system varies with time when stimulated with a zero-width infinite amplitude pulse.

initial time delay gap—Abbreviated ITD, the time in milliseconds (msec) between the arrival of the direct sound at a listener and the arrival of the first significant reflection. A reflection's significance is dependent upon its level in dB compared to surrounding scatter and its time interval. It is the first total spectrum reflection containing substantial energy relative to the direct sound.



intensity (sound)—(sound energy flux)—in a specified direction at a point is the average rate of sound energy transmitted in the specified direction through a unit area normal to this direction at the point.

inverse square law rate of level change—An attenuation of 6 dB for each doubling of distance from a source of sound.

L_{dn} (noise measurement)—A 24-hour Leq, except, 10 dB is added to all levels measured between 10:00 PM and 7:00 AM to account for the need for more quiet during sleep hours.

L_{den} (noise measurement)—A 24-hour Leq, except, 5 dB is added to all levels measured between 7:00 PM and 10:00 PM and 10 dB is added to all levels measured between 10:00 PM and 7:00 AM to account for the need for more quiet during sleep hours.

Leq (noise measurement)—Equivalent continuous sound level. The steady level which would produce the same sound energy over a stated period of time as the specified time-varying sound. Useful for studying long-term trends in environmental noise. A single number is used to define an entire measurement session.

Ln (noise measurement)—The level exceeded N% of the time, e.g. L90, the level exceeded 90% of the time, is commonly used to estimate ambient noise level.

level—The degree of intensity in dB of an audio signal.

liveness—A subjective description of a room related primarily to the average reverberation time of the middle octaves centered at 500 and 1000 Hz and to the balance between the direct and reverberant sound levels. It is also related to the volume of the room relative to the audience area.

lowpass filter—A filter that passes frequencies below a certain frequency and attenuates frequencies above that same frequency. A high-cut filter.

mean free path—The average distance traveled by sound between successive reflections.

near field—That part of a sound field, usually within about two wavelengths from a sound source, where there is no simple relationship between sound level and distance.

NC curve(s)—Noise criteria curves.

modulation transfer function—A measure of how well the amplitude modulation (variation of intensity with time) of a signal is preserved when the signal is sent through a particular transmission chain. Research has shown that a good portion of the intelligence in

human speech is contained in the modulation of the speech waveform. Preservation of the speech modulation patterns is important to maintain high intelligibility. Noise, echoes, and reverberation are found to decrease the effective modulation of the speech waveform and hence impair intelligibility.

See: STI and RASTI

Nyquist display—A plot of the tip of a vector that is changing in both length and angle as the frequency sweeps. The length of the vector is proportional to the magnitude of the energy, and the angle of the vector represents the phase of the signal. In 3-D space, the Nyquist Curve is like a corkscrew, or a spiral when viewed end-on, with the frequency axis pointing directly towards us.

“...real and imaginary components plotted as a rotating phasor. These are extremely useful in showing the partitioning of kinetic and potential energies frequency by frequency. Energy lying on the imaginary axis (vertical) is kinetic. Energy lying on real axis (horizontal) is potential energy. The ratio of imaginary to real is the ratio of kinetic to potential energy at that frequency.” *Don Davis*

octave—The interval between any two frequencies where the upper frequency is twice the lower frequency.
octave—the interval between any two tones whose frequency ratio is 2:1.

offset, time offset—*see:* receive delay

off-axis—Not directly in front of a microphone or loudspeaker.

off-axis coloration—In a microphone, the deviation from the on-axis frequency response that sometimes occurs at angles off the axis of the microphone. The coloration of sound (alteration of tone quality) for sounds arriving off-axis to the microphone.

PFC (Phase Response)—Phase frequency curve. Phase versus frequency display.

peak—On a graph of a sound wave or signal, the highest point in the waveform. The point of greatest voltage or sound pressure in a cycle.

peak amplitude—On a graph of a sound wave, the sound pressure of the waveform peak. On a graph of an electrical signal, the voltage of the waveform peak.

period—The time between the peak of one wave and the peak of the next. The time between corresponding points on successive waves. Period is the inverse of frequency.

phase—Phase is the measure of progression of a periodic wave. Phase identifies the position at any instant which a periodic wave occupies in its cycle. Phase describes the progress of a waveform in time relative to some starting point. If amplitude is plotted perpendicular to a time axis, phase may be represented as a position along the time axis. When two sinusoidal signals of the exact same frequency track each other

exactly in time, reaching their maximum, minimum and zero values in synchronization, they are said to be in-phase. If they are not synchronized, then it is as if one signal is delayed with respect to the other and there is a phase difference. Phase is measured in degrees or radians. Phase is frequency and time dependent. Phase measurements are the most precise indicators of alignment.

phase cancellation, phase interference—The cancellation of certain frequency components of a signal that occurs when the signal is combined with its delayed replica. At certain frequencies, the direct and delayed signals are of equal level and opposite phase (180 degrees out of phase), and when combined, they cancel out. The result is a comb-filter frequency response having a periodic series of peaks and dips. Phase interference can occur between the signals of two microphones picking up the same source at different distances, or can occur at a microphone picking up both a direct sound and its reflection from a nearby surface. Phase cancellation also occurs when two time-offset speaker drivers play the same frequency.

phase shift—Phase difference in degrees of phase angle between corresponding points on two waves. It is the fraction of a cycle by which one of the waves would have to be moved along the time axis to make the two waves coincide. One quantity is considered leading or lagging the other by the phase difference.

pink noise—A test signal containing all frequencies (unless band-limited), with equal energy per octave. Pink noise is a test signal, used with real time analyzers, for equalizing a sound system to the desired frequency response, and for testing loudspeakers.

pitch—The subjective lowness or highness of a tone. The pitch of a tone usually correlates with the fundamental frequency.

polar pattern—The characteristic pattern of a microphone and loudspeaker. A graph of microphone sensitivity plotted vs. angle of sound incidence. Some examples of polar patterns are omnidirectional, bidirectional, and unidirectional. Subsets of the unidirectional pattern are cardioid, supercardioid, and hypercardioid patterns.

polarity—The positive or negative direction of an electrical, acoustical, or magnetic force. Two identical signals in opposite polarity are 180 degrees apart at all frequencies. Polarity is not frequency dependent.

post processing data—Processing measurement results after performing test sweeps.

precedence effect—The effect of two sounds, approximately 20 milliseconds apart, that are coming from two places but which we localize to be at the location of the earlier arriving sound.

pre-delay—Short for pre-reverberation delay. The delay (about 30 to 100 milliseconds) between the



arrival of the direct sound and the onset of reverberation. Usually, the longer the pre-delay, the greater the perceived room size.

propagation—The travel of sound waves through a medium.

pure waveform—A waveform of a single frequency; a sine wave. A pure tone is the perceived sound of such a wave.

Q —The ratio of the sound pressure squared at a distance r in front of a source to the sound pressure squared, averaged over all directions. A source that radiates equally in all directions (spherical source) has a directivity factor Q of 1. A hemispherical source has Q of 2; a source in a corner (which radiates into one-quarter of a sphere) has a Q of 4, etc.

RASTI—Rapid Speech Transmission Index expressed in a decimal range of 0.2 for “bad” to 1.00 for “Excellent.” This method of evaluating speech intelligibility is based upon the method of the Speech Transmission Index (STI). Perfect transmission of speech implies that the speech envelope at the listener’s position replicates the speech envelope at the speaker’s mouth. Speech intelligibility can be quantified in terms of the changes in the speech envelope as a result of noise and reverberation in the room. In TEF, an equivalent of the RASTI method is achieved by testing only at the 500 Hz and 2 kHz octaves.

RT₆₀ (Reverberation time)--The time in seconds for the reverberant sound field to decay 60 dB after the sound source is shut off. It is calculated by measuring the rate of decay over at least the first 25 dB to 30 dB of decay and extrapolating what the RT₆₀ would be if the decay continued at that rate.

rarefaction—The portion of a sound wave in which molecules are spread apart, forming a region with lower-than-normal atmospheric pressure. The opposite of compression.

receive delay—In TEF, the difference in time between the start of the sweep and when the analyzer starts looking for it.

reflection—The bouncing or return of a sound wave from an object larger than one quarter wavelength of the sound. When the object is one quarter wavelength or slightly smaller, it also causes diffraction of the sound (sound bending around the object).

refraction—The change in direction of a sound wave that occurs when sound passes from one medium to another (from air to glass, to air or through layers of air with different temperatures).

reinforcement—See: sound reinforcement

relative phase—The phase of one sine wave compared with another.

resolution—The amount of detail we are able to resolve, or see, in the quantity that we are measuring.

resolution, frequency—Amount of detail we are able to resolve, or see, in the frequency domain. Measuring with 1 kHz of resolution will smear any details that have a repetition in less than 1 kHz. The effect of poor resolution on a frequency response curve is to smooth it out and minimize peaks and valleys. If we wish to increase frequency resolution to its highest possibility, we would use a receiver of infinite bandpass and infinite time window and a transmitter of pure sine-wave signals. The receiver could then, at any time during the measurement, hear any single frequency, but would have zero time information.

resolution, time—The amount of detail you are able to resolve, or see, in the time domain. Increasing the time resolution (making the time window smaller) will have the effect of decreasing the space-window ellipsoid. This will decrease the frequency resolution, since the units of time and frequency are reciprocals of each other. In making TEF measurements, time and frequency resolutions are adjusted by selecting appropriate combinations of sweep rate and filter bandwidth. This limits us to those frequencies that develop at least one wavelength within that time span.

resonance—A maximum response to an applied frequency; a peak in the frequency response. Resonance exists between a body or system and an applied force if any small change in frequency of applied force causes a decrease in amplitude of the responding body.

resonance frequency—The frequency at which resonance occurs. Of a traveling wave, resonance frequency is the change in amplitude as the frequency of the wave approaches or coincides with a natural frequency of the medium.

reverberant sound—*See:* reverberant sound field.

reverberation—The persistence of sound in a room after the original sound has ceased. It is caused by multiple sound reflections (echoes) that decrease in intensity with time, and are so closely spaced in time as to merge into a single continuous sound, which, eventually, is completely absorbed by the inner surfaces of the room. The timing of the echoes is random, and the echoes increase in number as they decay.

An example of reverberation is the sound you hear just after you shout in an empty gymnasium. An echo is a discrete repetition of a sound, while reverberation is a continuous fade-out of sound. Artificial reverberation is reverberation in an audio signal created mechanically or electronically rather than acoustically.

reverberant sound field—A sound field made of reflected sounds in which the time average of the mean square sound pressure is everywhere the same and the flow of energy in all directions is equally probable. This requires an enclosed space with essentially no acoustic absorption.



reverberation time—*See:* RT_{60} (Reverberation time).

room modes—Frequencies at which sound waves in a room resonate (in the form of standing waves), based on the room dimensions.

room time—*See:* RT_{60} (Reverberation time).

root mean square—The effective dc voltage of an ac signal. The square root of the mean value of the squares of the instantaneous values of a varying quantity. In periodic variation, the mean is taken over one period.

Sabin—a unit of absorption equal to the absorption of 1 square foot of surface which is totally sound absorbent.

Schroeder integration of reverberation—An integration of reverberant data in which the last energy is integrated first and the initial arrival is integrated last, all of which is normalized by the total. The integration simulates the effect of taking many time measurements and averaging them together.

signal delay—(commonly, but inaccurately called time delay) The difference in arrival times between two signals. A signal delay is also a device for delaying a signal.

signal-to-noise ratio—The ratio in decibels between signal and noise. An audio component with a high signal-to-noise ratio has little background noise accompanying the signal; a component with a low signal-to-noise ratio is noisy.

sine wave—A wave following the equation $y = \sin x$, where x is degrees and y is voltage or sound pressure level. The waveform of a single frequency.

sones—a unit of loudness. It is defined as the loudness of a 1000 cycle tone 40 dB above threshold. A millison is one-thousandth of a sone and is often called the loudness unit.

sound—Energy that is transmitted by pressure waves in air or other materials and is the objective cause of the sensation of hearing. Longitudinal vibrations in a medium in the frequency range 20 Hz to 20,000 Hz.

sound absorption—The change of sound energy into some other form—usually heat—in passing through a medium or on striking a surface.

sound decay—The dying of sound energy to equilibrium.

sound intensity—The rate of flow of sound energy through a unit area in a specified direction. The watt per square meter is the unit of sound intensity.

sound level—a term applied to data taken on instruments which meet the specifications for sound level meters drawn up by the American National Standards Institute (ANSI).

sound level meter—an apparatus for estimating the equivalent loudness of noise by an objective method.

sound power—The total sound power in watts radiated by a source.

sound pressure level (SPL)—Sound pressure level, in decibels, of a sound is 20 times the log to the base 10 of the ratio of the pressure of this sound to the ref pressure Pref.

$$\text{dB SPL} = 20 \log (P/P \text{ ref.}),$$

where $P \text{ ref} = 0.00002 \text{ pascal}.$

The value of pref should always be stated. A common reference pressure used in connection with hearing and the specification of noise is 0.00002 pascals.

sound reflections—*See:* reflections

sound wave—The periodic variations in sound pressure radiating from a sound source.

space window—An ellipsoid space around the speaker and microphone, inside of which sound reflections are included in the measurement. The speaker and microphone are at the foci of the ellipsoid. Sound reflections originating at the edge of the space window are attenuated 3dB and more distant reflections are attenuated by greater amounts.

On the TEF analyzer, the space window is determined by setting the bandwidth and sweep rate. For example, a 10-foot space window corresponds to a bandwidth of 88.5 Hz at a sweep rate of 10,000 Hz/second at a sound velocity of 1130 feet per second at room temperature. Formula: $B = SD/C$ where B = bandwidth of the tracking filter in Hz

S = sweep rate in Hz/sec

D = space window in feet

C = speed of sound, 1130 feet/sec

The larger the space window, the lower the frequency that can be measured accurately. That is, the frequency resolution increases as the space window increases. Therefore a relatively large, empty room is needed for low-frequency measurement.

speech intelligibility—A measure of sound clarity that indicates the ease of understanding speech. It is a complex function of psychoacoustics, signal-to-noise ratio of the sound source, and direct-to-reverberant energy within the listening environment. It is a function of signal level even in the presence of noise or reverberation. Intelligibility is at a maximum with sound pressure levels from about 70 to 90 dB, with a small decline in intelligibility at higher levels. Below 70 dB, intelligibility begins to decline, and it falls off rapidly with sound pressure levels below about 40 dB.



spectrum—The distribution of effective sound pressures or intensities measured as a function of frequency in specified frequency bands; the display of a signal in the frequency domain. The output vs. frequency of a sound source, including the fundamental frequency and overtones.**speed (of sound)**—1130 feet per second at 20 degrees centigrade.

specular reflections—Mirrorlike reflections of sound from a flat surface. Reflections that do not spread out.

speed (of sound)—in air, 1130 feet per second at 20 degrees centigrade.

speech reinforcement—The use of a sound system in an environment to increase speech intelligibility or sound power level.

SPL—*See*: sound pressure level.**standing wave**—An apparently stationary waveform, created by multiple reflections between opposite room surfaces. At certain points along the standing wave, the direct and reflected waves cancel, and at other points the waves add together or reinforce each other. These are sometimes called room modes.

start frequency—The starting frequency of a sweep. Traditionally noted in equations as F1.

stop frequency—The ending frequency of a sweep. Traditionally noted in equations as F2.

STI Speech Transmission Index—A single number that indicates the effect of a transmission system on speech intelligibility. A full STI test is accomplished by measuring seven individual one-second time-span ETC's at each of seven octave center frequencies between 125 Hz and 8 kHz. After each ETC test, the modulation transfer function (MTF) is calculated and the STI in each octave band is computed. The TEF test generator level at each octave band is adjusted to match the average spectral content of speech. At the conclusion of the test, the overall STI value is computed by taking a weighted average of the individual octave band STI values. *see* : speech intelligibility.

STC—Standard Transmission Class. A single number rating for describing sound transmission loss of a wall or partition.

sweep rate—The rate in Hz/second of a TEF sweep. It is the measure of how fast frequency is changing with respect to time.

sweep time—The duration of a TEF sweep.

swept sine wave—A sine wave made to vary uniformly in frequency from low to high or high to low. A frequency-modulated sine wave.

TDS time delay spectrometry—A method, conceived by Richard Heyser, that permits a spectrum that has been delayed to be measured with the signal delay removed. TDS measures in the frequency domain, then transforms the results mathematically for interpretation in the energy, frequency or time domains. In general terms, TDS is a way to measure energy passing through a system. TDS measurements describe what effect the system has on the energy passing through it. The results tell a great deal about the energy as well as the system. The principal advantages of TDS measurements are superior noise and distortion rejection properties, fast data gathering capability, and the ability to make acoustical measurements under actual use situations. In addition, TDS measurements easily handle test situations in which signal delays and nonlinearity are an inherent part of the system. Accuracy in TDS measurements depends on accurate measurement of both energy and time. Time delay spectrometry measurements include the frequency response, phase response, and time response data associated with other techniques, plus energy-time curves, and energy-time-frequency curves (3-D graphic display.)

TEF cube—A metaphor for envisioning how TEF displays time, energy, and frequency data.

3-D display—In TEF measurements, the 3-D display shows the change in magnitude/frequency response versus time for a number of individual TDS sweeps. Each sweep is offset in time by a constant amount, and on the screen form a three dimensional surface display. The three dimensions are time, energy and frequency.

time delay gap—A signal delay. The subjective judgment by a listener of how live or dead a room is does not depend on the reverberant sound field but rather on the initial time delay gap and the ratio of direct sound level to the early reflection level. *See:* initial time delay

time domain—In TEF measurements, that portion of the “TEF cube” in which time is the independent variable. Time domain measurements are made with time running horizontally along the axis.

time resolution—*See:* resolution, time.

time span—The time during which we listen for the effects of the signal on the device under test, and vice versa. It is shown in TEF (ETC) measurements as the amount of time on the X axis on the screen. It is dependent on the frequency span of the sweep and the number of data points displayed.

time window—A range of time over which signals are accepted by the analyzer. The relation between time window, bandwidth, and sweep rate is $T = B/S$, where T = width of time window in seconds; B = bandwidth in Hz, and S = sweep rate in Hz/sec. *see:* space window

two-port measurement—Measurement of a system by comparing its output signal to its input signal.

velocity—distance traveled, multiplied by the time elapsed.

wavelength— A wavelength is the distance traveled by a wave in a time of one cycle. The distance measured along the direction of propagation between two points which are in phase on adjacent waves. Low frequencies have long wavelengths; high frequencies have short wavelengths.

waveform—A graph of a signal's sound pressure or voltage vs. time. The waveform of a pure tone is a sine wave.

weighted—Referring to a measurement made through a filter with a certain specified frequency response. An A-weighted measurement is taken through a filter that simulates the frequency response of the human ear at low levels.

Appendix G: Best Frequency Resolution and the TEF Resolution V

Farrel M. Becker

Turning on *Best Frequency Resolution* in the TDS module's *Frequency Parameters* menu causes the computer to automatically set the TDS parameters such that your measurements will always have the best possible frequency resolution for the sweep time that you have selected. The longer the sweep time, the better the frequency resolution will be. To understand why this is so, let's review some terms and then look at what we call the *TEF Resolution V*.

The frequency resolution of a measurement determines the lowest frequency that we can measure with accuracy as well as how much detail we can see. A lower number yields better resolution. If we make a measurement with a frequency resolution of 1000 Hz, then any feature (a notch or bump) that is less than 1000 Hz wide will not be fully resolved. We will not see it accurately. It may still show up, but it may appear smoother than it really is. If we change the frequency resolution to 500 Hz (lower number therefore better resolution), we will get a clearer image.

Remember that as we increase the frequency resolution (smaller number) the *time* resolution decreases (bigger number). So while we have better frequency resolution and can see more detail, we are no longer able to reject reflections quite as well. In practice we must always

find a happy compromise between the time and frequency resolutions.

As you change the frequency resolution in the *Frequency Parameters menu*, you will notice that the Bandwidth value changes as well. Remember, that the frequency resolution is *note* equal to the bandwidth of the filter. In the TDS process, the bandwidth of the sweeping filter along with the speed at which it sweeps—the sweep rate—determines the resolution of the measurement. For TDS the frequency resolution (R_f) is equal to the sweep rate (SR) in Hz/s (hertz per second) divided by the bandwidth (BW) in Hz:

$$R_f = SR / BW$$

As you change the resolution, the computer calculates a new bandwidth that will yield the resolution you requested for the current sweep rate. (The sweep rate being determined by the Start Frequency, Stop Frequency and Sweep Time). The better the frequency resolution (smaller number), the larger the bandwidth.

Intuitively, you would think that to increase the frequency resolution you would have to *reduce* the bandwidth of the filter. This would allow you to look at a narrower portion of the spectrum. This is true for conventional swept spectrum analysis but not for TDS. Let's look at what goes on as you change the bandwidth of the filter.

In Illustration G-1 the TEF Resolution V is a graph, in the shape of the letter V, that shows how the frequency and time resolutions vary with the bandwidth of the sweeping filter and a fixed sweep rate. The vertical scale of the graph shows frequency resolution in Hz on the left side and time resolution in seconds on the right side. The horizontal scale is the bandwidth of the filter. Note that the horizontal scale is logarithmic. This is done so the V shaped curve will appear to be symmetric about the center of the graph. The bandwidth at the center of the horizontal scale is equal to the square root of the sweep rate and is marked $\text{Sqrt}(\text{SR})$. This particular graph uses a sweep rate of 10,000 Hz/s. Therefore, the bandwidth at the center of the horizontal scale is 100 Hz—the square root of 10,000. Bandwidth values to the left of center are less than the square root of the sweep rate and values to the right are greater.

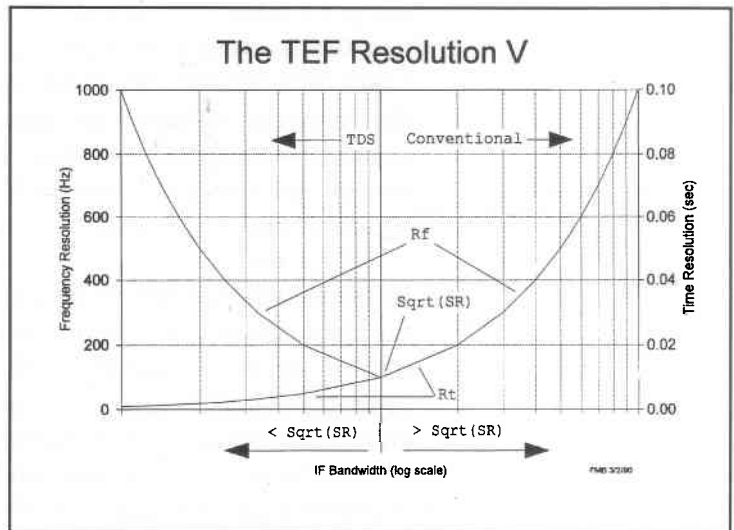


Illustration G-1
The TEF Resolution V

The frequency resolution is shown by the V shaped curve marked Rf that starts in the upper left corner, curves down to the center of the graph and then curves back up to the upper right corner. The time resolution is shown by the curve marked Rt and follows an exponential path (because of the logarithmic horizontal scale) from the lower left side of the graph to the upper right (actually lying directly below the frequency resolution curve to the right of the square root of the sweep rate). This tells us that the time resolution decreases (number gets bigger) linearly as the bandwidth increases.

The frequency resolution however behaves differently. As you can see in the graph, if we start with a bandwidth that is less than the square root of the sweep rate, the frequency resolution will *increase* (smaller number) as we increase the bandwidth until we reach the point where the bandwidth is equal to the square root of the sweep rate. As we continue to increase the bandwidth beyond the square root of the sweep rate, the frequency resolution *decreases*. The *Best Frequency Resolution* is equal to the square root of the sweep rate and occurs where the bandwidth is also equal to the square root of the sweep rate! Why?

Notice that the left side of the graph is labeled TDS and the right side is labeled Conventional with the square root of the sweep rate being the dividing line. This indicates that as long as the bandwidth is less than or equal to the square root of the sweep rate we are doing TDS. If however, we set the bandwidth to a value that is greater than the square root of the sweep rate, we are no longer doing TDS but are instead performing

conventional swept spectrum analysis. On the TDS side we have the advantage of a time resolution that is the reciprocal of the frequency resolution (the frequency resolution is equal to 1 divided by the time resolution) and allows good time selectivity. On the conventional side, while we can still get the same frequency resolutions that we can on the TDS side, but we cannot get good time resolution.

By turning on *Best Frequency Resolution* in the *Frequency Parameters menu*, you automatically set the bandwidth to the square root of the sweep rate and obtain the best possible frequency resolution for the current sweep rate.

Suppose the *Best Frequency Resolution* isn't good enough. How can you get a better frequency resolution than the "Best?" You simply increase the sweep time. As you increase the sweep time, and therefore decrease the sweep rate, the bandwidth will automatically be reset to the square root of the new, slower sweep rate yielding a new "better" frequency resolution..

When should you use *Best Frequency Resolution*? Generally, for acoustic measurements, loudspeaker frequency responses, etc., you would not use it. Looking at the graph you can see that when you have a very high frequency resolution (small number) the time resolution is very poor (large number). This usually allows reflections into the measurement. So, for acoustic measurements, you will generally set the time/frequency resolutions to reject any reflections.

For electronic measurements, frequency responses of equalizers, loudspeaker impedances, etc., you will almost always want to use *Best Frequency Resolution*. You always want the *Best Frequency Resolution* you can get. In acoustic measurements we are limited by the arrival of reflections. In electronic measurements there are no reflections! Now we can turn *Best Frequency Resolution* on and set the sweep time to get the frequency resolution that we want. This way we get the desired frequency resolution with the shortest possible sweep time.

One final note on sweep rate and bandwidth. The current values are always shown at the bottom of the *Frequency Parameters menu*. You can set them manually if you want to. There really is no reason to do so however. What we are really interested in is the time/frequency resolution that results from the sweep rate and bandwidth. Set the sweep time and resolution that you want and let the computer do the work of setting the sweep rate and bandwidth.

Bibliography

The following bibliography references materials on the subjects of acoustics, measurement, time delay spectrometry, perception, and recording techniques.

Glen M. Ballou., ed. *Handbook for Sound Engineers. The New Audio Cyclopedia.* Carmel, Indiana: Howard W. Sams, 1991.

Bruce Bartlett. *Introduction to Professional Recording Techniques.* Carmel, Indiana: Howard W. Sams, 1987.

Leo L. Beranek. *Acoustical Measurements, Rev. ed.,* Cambridge, Massachusetts: Published for the Acoustical Society of America by the American Institute of Physics. 1988.

Lothar Cremer, Helmut A. Miller, Theodore J. Schultz. *Principles & Applications of Room Acoustics.* Essex, England: Applied Science Publishers, Ltd., 1978.

Malcom J. Crocker. *Noise Control.* New York, New York: Van Nostrand Reinhold Co., 1982.

Don Davis and Carolyn Davis. *Sound System Engineering.* Indianapolis, Indiana: Howard W. Sams, 1989.

J.R. Hassail, and K Zaveri. *Acoustic Noise Measurements*. Nærom, Denmark: K. Larsen & Sons, A/S. 1979.

Richard C. Heyser. *Time Delay Spectrometry, An Anthology of the Works of Richard C. Heyser on Measurement, Analysis, and Perception*. New York, New York: Audio Engineering Society, Inc., 1988.

Peter Mapp. *The Audio System Designer Technical Reference* by Klark-Teknik Plc.

Harry B. Miller. *Acoustical Measurements, Volume 16, Methods and Instrumentation*. Stroudsburg, Pennsylvania: Hutchinson Ross Publishing Co., 1982.

Michael Rettinger. *Acoustic Design & Noise Control, Volume 1*. New York, New York: Chemical Publishing Co., Inc., 1977.

Earl D. Schubert. *Psychological Acoustics, Volume 13*. Stroudsburg, Pennsylvania: Dowden, Hutchinson & Ross, Inc., 1979.

Index

Symbols

(more) 4-2

3D (Waterfall)

Auto Scaling 4-55

Bottom of Scale 4-55

Curve order 4-55

display 4-43

measurements 5-27

Parameters 4-30

Perspective 4-55

Top of Scale 4-55

3D End Delay

cursor button 6-29

3D Start Delay

cursor button 6-29

A

About 4-1, 4-9

acoustic

center,origin F-1

action buttons 3-7

Add Noise to STI RASTI 4-59

Adjust colors 4-66

ALcons 6-25

definition F-3

measuring with ETC 6-34

setting the cursor 6-30

Alt-Beep 4-72

Alt-Brief 4-3

Alt-Close 3-7, 4-3

Alt-Full 4-3

Alt-Info 4-3

Alt-Mouse 4-72

Alt-Print 4-3

Alt-Undo 3-7, 4-15

ambient noise F-1

amplitude, peak F-15

analyzer F-2

anechoic F-3

Arrow keys 3-2

articulation loss of consonants. *See*

ALcons

ASCII

3D file format E-25

ETC file format E-2

NC ASCII files E-6

RASTI file format E-13

Saving TDS files as 4-5

TDS file format E-9

attenuate F-3

Auto Repeat 4-13

Auto Scaling 4-45, 4-69

3D (Waterfall) 4-55

Frequency Response 4-48, 4-49, 4-50, 4-51

Time Response 4-44

Average 4-13

Averaging

Magnitude 4-38

Measurement type 4-36

Number of Sweeps 4-36

Pause Between Sweeps 4-36

Test Sequencing 4-37

Vector 4-38

B

Back to Front, 3D 4-56, 4-71
band-pass filter F-3
Bandwidth
 (3D) 4-32
 (TDS) 4-22, 4-25
 definition F-3
 Optimum 4-21
 sweep rate B-10
Base address 4-82
Baud Rate 4-81
Best Frequency Resolution
 (3D) 4-31
 (TDS) 4-21, 4-25
Binary, saving TDS files as 4-5
Bottom of Scale
 3D (Waterfall) 4-55
 Frequency Response 4-48,
 4-49, 4-50
 Time Response 4-44, 4-45

C

Calibration 4-76
 equipment arrangement 5-15
 talker/speaker 5-14
Center Frequency (ETC) 4-16
Channel 4-74
Clear All 4-5
Colors 4-66
COM port 2-3
COM1 4-81
COM2 4-81
comb filter F-4
commands 3-1

Communication 4-81
complex wave F-4
compression F-4
Configuration 4-1, 4-7
coverage F-4
coverage angle F-4
crest factor F-4
critical
 distance F-4
 frequency F-4
Cursor 4-63
 relative 4-65
 to examine data 6-11
Cursor is OFF D-2
Curve Order
 3D (Waterfall) 4-55, 4-56, 4-71
Customer support 1-5

D

data
 display options 4-41
 interpretation C-1
 storage
 disk, directory 4-4
dB. *See* decibel
DC Shift Trigger 4-34
deadness F-5
decay rate F-5
decay time F-5
decibel F-5
delay F-5
destination drive:\path 2-2
Difference 4-41, 4-61
diffraction F-6
diffuse field F-6

diffuser F-6
diffusion F-6
direct sound F-6
Direct to Reverberant Energy
 Ratio 6-29
disk drives. *See* drive and directory
display, calibrating 6-5
Display menu 4-41
 3D (Waterfall) 4-55, 4-56
 Add Noise to STI RASTI 4-59
 Adjust colors 4-68
 Alt-Beep 4-72
 Alt-Mouse 4-72
 Cursor 4-63
 Difference 4-61
 Frequency Response 4-42,
 4-47, 4-48, 4-49, 4-50, 4-51
 NC 4-57
 Overlay 4-60
 RT60 cursor 4-66
 STI 4-58
 Time Response 4-42, 4-44, 4-46
Display Rotation, Nyquist 4-51,
 4-70
Distance resolution
 (3D) 4-31
 (TDS) 4-21
Distance Unit 4-77, 4-80, 6-6
Do 3D Test 4-13
Do Average 4-13
Do Frequency Test 4-11
Do Noise Test 4-11, 4-43
Do RASTI Test 4-12
Do STI Test 4-12
Do Time Test 4-11
Doing Noise Test D-3

domain F-6
doppler effect F-6
Drive & Dir 4-1, 4-4
drive and directory
 changing 4-4
 default 2-2

E

early decay time 6-26, 6-28,
 6-33, F-7
early RT60 5-24
early sound F-7
Echoes A-7, F-7
edit field 3-5
EE, EEPROM 4-1
EFC F-7
ellipsoid B-12, F-7. *See also* space
 window
End Receive Delay
 (3D) 4-32, 5-28, 5-30, 6-29
Energy Frequency Curve F-7
Enter key 3-7
Equipment arrangement
 ETC 6-4
 STI calibration 5-15
 STI/RASTI tests 5-18
 TDS 6-4
Erase 4-1, 4-7
Error codes. *See* Messages
Escape key 3-7
ETC
 definition F-8
 display 6-3
 file format E-2
 measurement 5-2
Extensions 4-6

F

- far field F-8
- FFT B-4, F-8
- File format
 - 3D files E-25
 - ETC ASCII files E-2
 - NC ASCII files E-6
 - STI and RASTI E-13
 - TDS ASCII files E-9
- File menu 4-1
 - About 4-9
 - Configuration 4-7
 - Drive & Dir 4-4
 - Erase 4-7
 - Go to 4-9
 - Open Old 4-2
 - Printer settings 4-8
 - Quit 4-9
 - Save as 4-4
- file size 4-3
- filenames, extensions 4-2, 4-6
- filter
 - bandpass F-3
 - highpass F-11
 - lowpass F-13
- flutter echo F-8
- focused reflections F-8
- Fourier transform B-4, F-9
- free field F-9
- Frequency
 - Center (ETC) 4-16
 - defined A-4
 - Limits, Start and Stop 4-17
 - Resolution F-9, F-20
 - (3D) 4-31
 - (TDS) 4-21
 - Best (TDS) 4-21
 - response F-10
 - Span B-9, F-10
 - (ETC) 4-17
 - Start F-26
 - (3D) 4-30
 - (ETC) 4-16
 - (TDS) 4-20
 - Stop F-26
 - (3D) 4-30
 - (ETC) 4-16
 - (TDS) 4-20
- Frequency Response
 - (TDS) Parameters 4-20
 - Bottom of Scale 4-51
 - Display
 - Heyser Spiral 4-42
 - Magnitude 4-42
 - Nyquist 4-42
 - Phase 4-42
 - Graph, Nyquist 4-51
 - Nyquist 4-51
 - Test 5-6
 - Top of Scale 4-51
- Front to Back, 3D 4-56
- FTC F-10
- full scale time F-10
- function keys 3-7
- fundamental F-10

G

Gain definition F-10
Go to command 3-8, 4-9
Graph
 Nyquist 4-51
 Size 4-8

H

Hamming window 4-18
harmonic F-10
Help from Techtron 1-5
hertz F-11
heterodyning F-11
Heyser Spiral
 Frequency Response 4-53
 Time Response 4-42, 4-46
Heyser spiral display 4-53
HI base address 2-4
HI communication 4-82
highpass filter F-11
Horizontal Scale
 3D (Waterfall) 4-55
 Frequency Response 4-48,
 4-49, 4-50
 Linear, Log 4-48
Host Interface 2-3
Hz—Abbreviation F-11

I

impulse response F-11
input hardware, setting 6-5
Input menu 4-73
 Calibration 4-76
 Distance Unit 4-80
 Propagation Speed 4-79

Reference Unit 4-78
Volts per Reference Unit 4-78
Zero dB Reference Value
4-78

Communication 4-81

Baud Rate 4-81
HI Base 4-81
Port 4-81

Settings 4-74

Channel 4-74
Input 4-74
Loopback 4-74

Installing 2-1

Integration Time, Noise Test 4-27
interrelated parameters 4-20, B-8

J

job description, printing 4-3

K

Keyboard, test sequencing 4-34

L

Lden F-12
Ldn F-12
Leq F-12
level F-13
Line Output, Noise Test 4-27
Linear regression line, RT 60 6-29
linear scale 4-48
liveness F-13
Ln F-13
logarithmic scale 4-48
Loopback 4-74
loudspeakers, measuring 5-28
lowpass filter F-13

M

Magnitude. *See also* Frequency
response: magnitude

Averaging 4-38

Measure menu 4-11

Do 3D Test 4-13

Do Average 4-13

Do Frequency Test 4-11

Do Noise Test 4-11

Do RASTI Test 4-12

Do STI Test 4-12

Do Time Test 4-11

Measure Noise, STI/RASTI 4-29

Measurements

3D 5-28

ALcons 6-25

first time 6-1

Frequency Response 5-6

RT60 6-25

STI/RASTI 5-15

Time response 5-2

two-port F-29

menu title

selecting 3-3

Menus

Display 4-41

File 4-1

Input 4-74

Measure 4-11

Parameters 4-15

Message boxes 3-8

Messages

Awaiting the Trigger D-1

Bandwidth D-1

Can not save +1 D-1

Can't difference D-1

Check communications D-2

Cursor is off D-2

Delay before test ## seconds
D-2

Disk error D-2

Doing noise test D-3

Drive not ready D-3

Error file... D-3

Error in data file D-3

Error writing D-3

Files not found D-5

Insufficient disk space D-4

Invalid path D-4

Loading... D-4

Measure halted D-4

Measurement interrupted D-4

Min = 0 Max = 60 D-4

No data available D-6

No data current D-5

Not enough memory D-6

Obtain STI data D-5

Parameters changed D-7

Path/filename exists D-5

Performing fft D-7

Receiving data D-7

RT60 out of range D-7

Start frequency should be D-8

Test Interrupted D-8

Test Timed Out D-8

Testing D-8

The printer is not ready D-8

The TEF is not responding! D-9

There is not data ... D-9
Time span should not exceed
sweep time D-9
modulation transfer function 5-25
modules, changing 4-9
mouse 3-2, 4-41
MTF. *See* modulation transfer
function

N

NC (Noise)
about the display 5-11
curves 5-12
display 4-43, 4-57
file format E-6
measurements 5-10
Parameters 4-28
rating 5-12
table or graph 4-57
near field F-13
Noise
adding to STI/RASTI 4-41
ambient F-1
pink F-17
signal-to-noise ratio F-23
Noise criteria measurements. *See*
NC (Noise)
noiseless test 4-12
Number of Curves (3D) 4-33
Number of Samples
(3D) 4-33
(ETC) 4-17
(TDS) 4-22
Nyquist F-14

Frequency Response 4-51

O

octave F-14
Octave smoothing 4-51, 4-70
Frequency Response 4-49,
4-50, 4-51
Nyquist 4-51
off-axis F-15
On-screen messages D-1
Open Old 4-2
Output
Pink noise 4-39
Port 4-8
Sine Wave tone 4-39
White noise 4-39
Overlap ratio 5-31
Overlay 4-41, 4-60
overload 4-75, 5-11

P

Parameters 4-15
guidelines
Frequency Response 4-24
STI 5-16
Time Response 4-18
interdependent 4-15
Menu
3D test 4-30
Averaging 4-36
Frequency Response 4-20
Noise Test 4-27
STI/RASTI 4-29
Time Response 4-16, 6-3
parent directory 4-2

Path, changing 4-4
 Pause
 Between Curves (3D) 4-33
 Between Sweeps, Averaging 4-36
 peak amplitude F-15
 percent ALcons. *See* ALcons
 Performing FFT D-7
 Perspective, 3D 4-55, 4-71
 Phase. *See* Frequency Response:
 Phase
 cancellation F-16
 definition F-15
 interference F-16
 relative F-19
 shift F-16
 Phase and phase shift A-4
 Pink Noise 4-39, F-17
 polar pattern F-17
 polarity F-17
 Port 4-81
 post processing data F-17
 pre-delay F-17
 Preamp Gain 4-75
 precedence effect F-17
 Print 4-1, 4-8
 Printer
 Graph Size 4-8
 Resolution 4-8
 Settings 4-1, 4-8
 printing
 job description 4-3
 propagation F-18
 Propagation Speed 4-77, 4-79, 6-6

pull-down menu 3-1
 Pulse trigger, test sequencing 4-34
 pure waveform F-18

Q

Q F-18
 Quick keys 3-2
 Quit 4-9

R

rarefaction F-19
 RASTI
 converting to ALCONS 5-24
 definition F-18
 file format E-13
 Receive Delay B-9
 (ETC) 4-17
 (TDS) 4-22
 definition F-19
 setting with phase 6-19
 Start (3D) 4-31
 Step (3D) 4-32, 5-30
 Receiving Data D-7
 Reference Unit 4-77, 4-78, 6-6
 Reflections A-7
 definition F-19
 eliminating B-2
 finding 6-12
 focused F-8
 refraction F-19
 reinforcement F-19
 speech F-26
 relative cursor 4-65, 6-12
 relative phase F-19
 required files 2-6

Resolution
 (3D) 4-31
 Best Frequency
 (3D) 4-31
 (TDS) 4-25
 Distance
 (3D) 4-31
 (TDS) 4-21
 Frequency C-5, F-9, F-20
 (3D) 4-31
 (TDS) 4-21
 measurement B-5
 message D-7
 printing 4-8
 Time F-20
 (3D) 4-31
 (TDS) 4-21
 resonance F-20
 response
 frequency F-10
 impulse F-11
 phase F-15
 reverberation A-9, F-21
 Reverberation time 6-26, A-11,
 F-19
 room
 measuring with 3D 5-30
 modes A-11, F-22
 time F-22
 root mean square F-22
 RT60 6-31, 6-32, F-19
 calculation 6-35
 cursor 4-66, 4-72
 cursor button

3D End Delay 6-29
 3D Start Delay 6-29
 Slope 6-29
 TDS delay 6-29
 early decay time 6-28
 early RT60 in STI 5-24
 Linear regression line 6-29
 measurements 6-26
 out of range message D-7
 Schroeder curve 6-28
 setting the cursor 6-28
 RT60 cursor is on...off D-7

S

Sabin F-22
 Save +1 4-6
 Save As 4-1, 4-4
 Schroeder integration 6-28, F-22
 serial communication 4-81
 Serial port 4-81
 signal delay. *See* receive delay
 sine wave F-23
 Sine Wave Tone 4-39
 SL directory 2-6
 command tree 2-6
 SL.BAT 2-6
 Slope cursor button 6-29
 sone F-23
 sound
 absorption F-23
 decay F-23
 definition F-23
 discrete arrivals F-6
 early F-7
 intensity F-23

- level F-24
- near field F-13
- power F-24
- reverberant field F-21
- speed of F-26
- sound level meter F-24
- Source Calibration, STI/RASTI 4-29
- space window B-12, B-14,
F-24. *See also* ellipsoid
- span, frequency F-10
- Speaker EQ, testing 5-17
- specular reflections F-26
- speech intelligibility F-25
- Speech Transmission Index F-27
- SPL F-24
- Standard Transmission Class F-27
- Standing waves A-12, F-26
- Start Frequency
 - (3D) 4-30
 - (ETC) 4-16
 - (TDS) 4-20
- Start Receive Delay
 - 3D Tests 6-29
 - 3D tests 5-28, 5-30
- STI F-27
 - about the measurement 5-25
 - Add Noise to STI RASTI 4-59
 - display 4-43
 - graph 5-21, 5-22
 - table 5-21, 5-22
 - Early RT60 5-24
 - equipment arrangement 5-19
 - file format E-13
 - graph or table 4-58
 - noiseless test 4-12
 - score 5-24
 - Signal-to-noise ratio 5-24
- Stop Frequency
 - (3D) 4-30
 - (ETC) 4-16
 - (TDS) 4-20
- Sweep Rate B-8, F-27
 - (3D) 4-30
 - (TDS) 4-20
- Sweep Time F-27
 - (3D) 4-30
 - (ETC) 4-17
 - (TDS) 4-20
- Sweeping through zero B-9
- swept sine wave F-27

T

TDS

- definition F-28
- file format E-9
- measurements 5-6
- parameters 4-20
- TDS Delay cursor button 6-29
- Telephone numbers 1-5
- Test sequencing
 - Automatic 4-33
 - Averaging 4-37
 - DC Shift Trigger 4-34
 - Keyboard 4-34
 - Pulse trigger 4-34
- Testing D-8
- 3-D display F-28

Time

delay spectrometry F-28

delay gap F-29

domain F-29

offset F-14

Resolution

3D tests 4-31, 5-29

TDS 4-21, F-20

Response

(ETC) Parameters 4-16

Span F-29

(ETC) 4-17

calculating 4-18

window F-29

Time and frequency 6-3

Time Response

center frequency 6-7

display options 4-42

equipment arrangement 6-4

Heyser Spiral 4-46

Magnitude 4-44

parameters 6-1

set up, loudspeaker 5-2

uses for 6-1

viewing 6-3

toggle choices 3-6

Top of Scale 4-70

3D (Waterfall) 4-55

Frequency Response 4-49,
4-50, 4-51

Time Response 4-44

V

Vector averaging 4-38

velocity F-29

Volts Per Reference Unit 4-77, 6-6

W

warning beep 4-41

waveform A-1, F-30

wavelength F-30

weighted F-30

White Noise 4-39

Window

(ETC) 4-17, 4-18

space F-24

time F-29

Wrap

Phase 4-48, 4-50

Z

Zero dB Reference Value 4-77, 6-6

Zero sweep rate B-8